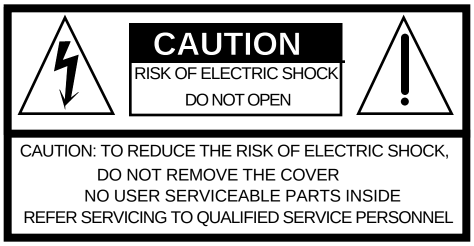
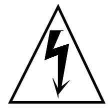
K2600

Musician's Reference

KURZWEIL
Music Systems

Part Number: 910331 Rev. A





The lightning flash with the arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.

The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.

IMPORTANT SAFETY & INSTALLATION INSTRUCTIONS

INSTRUCTIONS PERTAINING TO THE RISK OF FIRE, ELECTRIC SHOCK, OR INJURY TO PERSONS

WARNING: When using electric products, basic precautions should always be followed, including the following:

1. Read all of the Safety and Installation Instructions and Explanation of Graphic Symbols before using the product.
2. This product must be grounded. If it should malfunction or break down, grounding provides a path of least resistance for electric current to reduce the risk of electric shock. This product is equipped with a power supply cord having an equipment-grounding conductor and a grounding plug. The plug must be plugged into an appropriate outlet which is properly installed and grounded in accordance with all local codes and ordinances.

   DANGER: Improper connection of the equipment-grounding conductor can result in a risk of electric shock. Do not modify the plug provided with the the product - if it will not fit the outlet, have a proper outlet installed by a qualified electrician. Do not use an adaptor which defeats the function of the equipment-grounding conductor. If you are in doubt as to whether the product is properly grounded, check with a qualified serviceman or electrician.
3. **WARNING:** This product is equipped with an AC input voltage selector. The voltage selector has been factory set for the mains supply voltage in the country where this unit was sold. Changing the voltage selector may require the use of a different power supply cord or attachment plug, or both. To reduce the risk of fire or electric shock, refer servicing to qualified maintenance personnel.
4. Do not use this product near water - for example, near a bathtub, washbowl, kitchen sink, in a wet basement, or near a swimming pool, or the like.
5. This product should only be used with a stand or cart that is recommended by the manufacturer.
6. This product, either alone or in combination with an amplifier and speakers or headphones, may be capable of producing sound levels that could cause permanent hearing loss. Do not operate for a long period of time at a high volume level or at a level that is uncomfortable. If you experience any hearing loss or ringing in the ears, you should consult an audiologist.
7. The product should be located so that its location or position does not interfere with its proper ventilation.
8. The product should be located away from heat sources such as radiators, heat registers, or other products that produce heat.
9. The product should be connected to a power supply only of the type described in the operating instructions or as marked on the product.
10. This product may be equipped with a polarized line plug (one blade wider than the other). This is a safety feature. If you are unable to insert the plug into the outlet, contact an electrician to replace your obsolete outlet. Do not defeat the safety purpose of the plug.
11. The power supply cord of the product should be unplugged from the outlet when left unused for a long period of time. When unplugging the power supply cord, do not pull on the cord, but grasp it by the plug.
12. Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.
13. The product should be serviced by qualified service personnel when:
    - A. The power supply cord or the plug has been damaged;
    - B. Objects have fallen, or liquid has been spilled into the product;
    - C. The product has been exposed to rain;
    - D. The product does not appear to be operating normally or exhibits a marked change in performance;
    - E. The product has been dropped, or the enclosure damaged.
14. Do not attempt to to service the product beyond that described in the user maintenance instructions. All other servicing should be referred to qualified service personnel.
15. **WARNING:** Do not place objects on the product's power supply cord, or place the product in a position where anyone could trip over, walk on, or roll anything over cords of any type. Do not allow the product to rest on or be installed over cords of any type. Improper installations of this type create the possibility of a fire hazard and/or personal injury.

RADIO AND TELEVISION INTERFERENCE

WARNING: Changes or modifications to this instrument not expressly approved by Young Chang could void your authority to operate the instrument.

IMPORTANT: When connecting this product to accessories and/or other equipment use only high quality shielded cables.

NOTE: This instrument has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This instrument generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this instrument does cause harmful interference to radio or television reception, which can be determined by turning the instrument off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the instrument and the receiver.
- Connect the instrument into an outlet on a circuit other than the one to which the receiver is connected.
- If necessary consult your dealer or an experienced radio/television technician for additional suggestions.

NOTICE

This apparatus does not exceed the Class B limits for radio noise emissions from digital apparatus set out in the Radio Interference Regulations of the Canadian Department of Communications.

AVIS

Le present appareil numerique n’emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques de la class B prescrites dans le Reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada.

SAVE THESE INSTRUCTIONS

Young Chang Distributors

Contact the nearest Young Chang office listed below to locate your local Young Chang/ Kurzweil representative.

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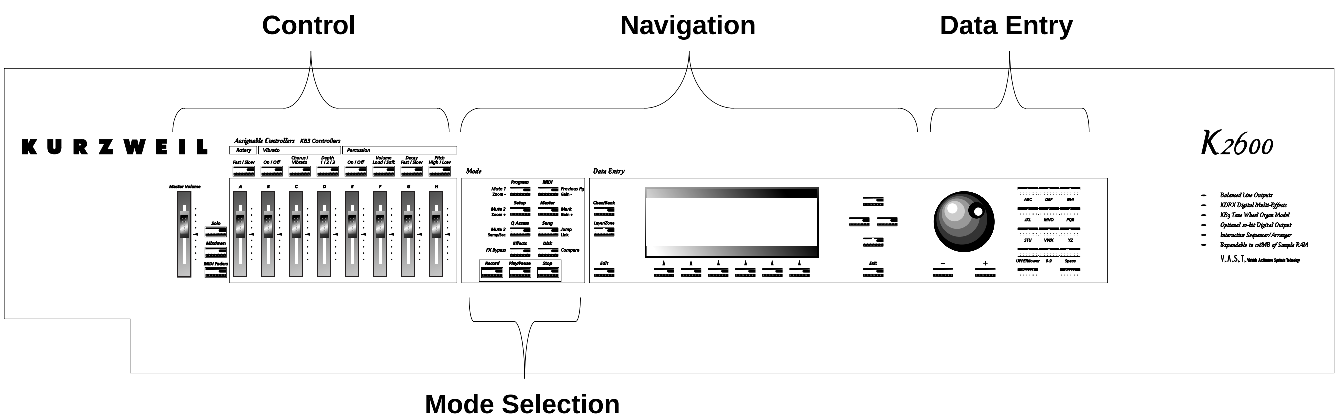
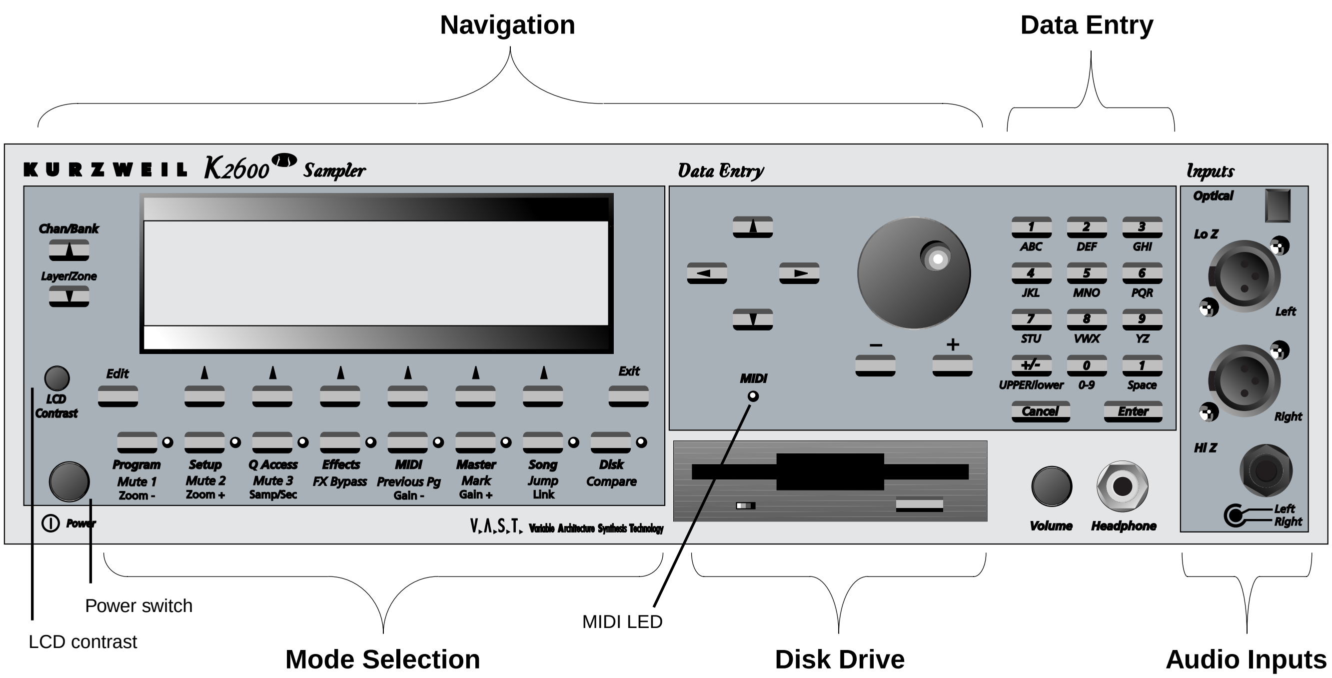
Chapter 1
Front Panel

Front Panel Quick Reference

This section describes features that, unless specified otherwise, are common to both the rack versions of the K2600 (K2600R and K2600RS) as well as the keyboard versions of the K2600 (K2600, K2600S, K2600X, and K2600XS). The buttons and sliders that are unique to the keyboard models are described on page 1-4.





Volume Knob/ Slider

Controls mixed audio outputs and headphone jack only. Does not send MIDI Volume (MIDI 07).

Mode Buttons

Press any of these eight buttons to enter the corresponding mode.

Chan/Bank Buttons

Scroll through the layers of the current program while in the Program Editor. Scroll through the zones in the current setup while in Setup mode. Scroll through the Quick Access banks while in Quick Access mode.

Edit Button

Functional in most modes. Press **Edit** to modify the currently selected object or parameter. If it's not editable, pressing **Edit** will do nothing.

There are editors available from every mode but Disk mode. The effect of pressing **Edit** in each of the modes is listed below.

When in this mode	Pressing the Edit button…
Program mode	…enters the Program Editor, where you can edit the currently selected program. Chapter 6 in the *Musician's Guide* covers the Program Editor.
Setup mode	…enters the Setup Editor, where you can edit the currently selected setup. Chapter 7 in the *Musician's Guide* describes the Setup Editor.
Quick Access mode	…enters the Quick Access Editor, where you can change the program or setup assigned to the bank slot that was selected when you entered the Quick Access Editor. See Chapter 8 in the *Musician's Guide*.
Effects mode	…if the Studio parameter is highlighted, enters the Studio Editor, where you can edit the currently selected studio. Chapters 9 and 15 in the *Musician's Guide* explain studios, the Studio Editor, FX presets, and the FX Preset Editor.
MIDI mode	…enters the Velocity Map or Pressure Map Editor if the Velocity or Pressure Map parameter is selected on either the TRANSMIT page or the RECEIVE page. See Chapter 18 in the *Musician's Guide*. Takes you to the Program Editor if the Program parameter is selected on the CHANLS page. See Chapter 6 in the *Musician's Guide*.
Master mode	…enters the Velocity Map, Pressure Map, or Intonation Table Editor if the VelTouch, PressTouch, or Intonation parameter is selected. See Chapter 18 in the *Musician's Guide*.
Song mode	…enters the Song Editor. The Song Editor is discussed in Chapter 12 in the *Musician's Guide*. Takes you to the Program Editor if the Program parameter is highlighted when **Edit** is pressed.
Disk mode	…has no effect.

Table 1-1 Navigating with the Edit Button

Soft Buttons

Functions change depending on current display page. Function of each button is displayed on bottom line of display.

Exit Button

Press to leave various editors. If you've made any changes while in the editor, you will be prompted to save them.

Cursor Buttons

Press the corresponding button to move the cursor up, down, left, or right in the display. Different parameter values will be highlighted as buttons are pressed.

Alpha Wheel

For data entry. Rotate clockwise to increase value of currently selected parameter, counterclockwise to decrease.

Plus / Minus Buttons (- and +)

Under the Alpha Wheel. Press to increase or decrease the value of the currently selected parameter by the smallest possible amount. Don't confuse this with the **+/-** button on the alphanumeric buttonpad.

Alphanumeric Buttonpad

For Numeric Characters

Enter the value numerically instead of using the Alpha Wheel or **Plus**/**Minus** buttons. Press **Enter** when finished. Press **Cancel** to restore a parameter to its previous value. Pressing **Clear** is equivalent to pressing **0** without pressing **Enter**.

For Alphabetic Characters

When naming objects, you can use the alphanumeric pad to enter letters instead of numbers. If you're renaming a program, for example, just position the cursor under the character you want to change, then press the corresponding numeric button, as labeled. Press the button as many times as necessary to enter the desired character. Pressing **Clear** will enter a space before the selected character. The **0** button will enter the numerals 0–9 when pressed repeatedly.

Here's an example. To enter the letter **C** in a blank space, press **1** three times. You can press the **+/-** button before or after entering the letter.

The **Cancel** button is equivalent to the soft button, and **Enter** is the same as **OK**. The **Clear** button replaces the currently selected character with a space. The **+/–** button toggles between uppercase and lowercase letters.

When you press the **+/–** button on the alphanumeric pad, the currently selected character (the one with the cursor under it) will switch from upper case to lower case, and vice versa. The **+/–** button is a toggle; that is, if you switch from lower to upper case, all further entries will be in upper case until you press the **+/–** button again.

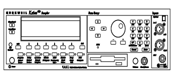
There are several punctuation characters available as well, but they can be entered only with the Alpha Wheel or **Plus/Minus** buttons. The punctuation characters are between **z** (lower case) and **0**.

Special Alphanumeric Buttonpad Functions

When you're in Quick Access mode, the Alphanumeric buttonpad can be used to select the entries in the current Quick Access bank. The layout of the alphanumeric buttonpad corresponds to the layout of Quick Access bank entries as seen on the Quick Access-mode page.

There's also a shortcut for selecting different QA banks while in QA mode. Just press the **+/–** or **Clear** button on the alphanumeric pad, and you'll be prompted to enter a bank number. Type the desired number on the alphanumeric pad, then press **Enter**. The bank will be selected, and you'll return to the Quick Access page.

You can also use the alphanumeric pad to select strings to search for in the currently selected list of objects, and to enter new strings to search for. The search function is described fully on page 3-8 of the *Musician's Guide*.



Lastly, rack users can play notes from the numeric keypad by holding down the **Cancel** button while pressing alphanumeric buttons. This is described fully on page 3-10 of the *Musician's Guide*.

The Display

You may want to adjust the contrast of the display for different lighting conditions. On keyboard models, the adjustment knob is on the rear panel, between the MIDI ports and the continuous controller pedal jacks. On rack models, it's on the front panel, above the power switch.

MIDI LED (Rack Models Only)

Lights when the K2600 is receiving MIDI information at its MIDI In port.

Special Keyboard Functions

This section describes the buttons and sliders that are unique to the keyboard models of the K2600. Features common to both rack and keyboard models are described starting on page 1-1.

Programmable controllers:

Sliders A–H, and the buttons above them,

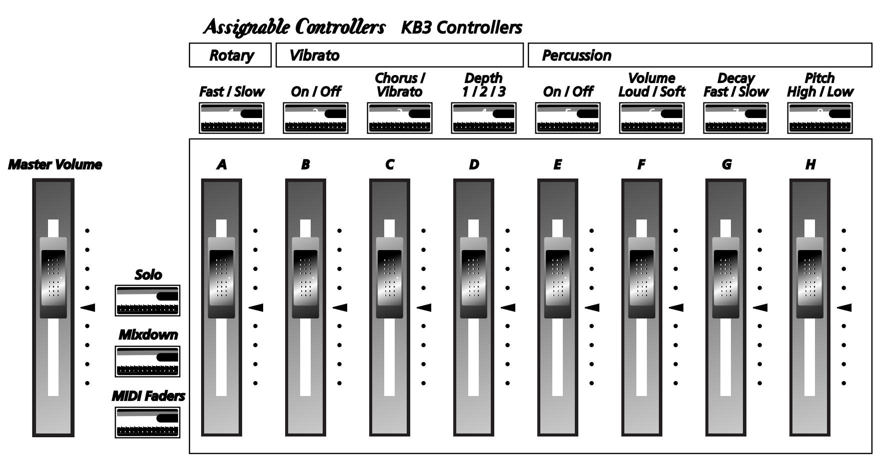
Pitch Wheel and Mod Wheel

Panel switches (Buttons 9 and 10)

Large and small ribbons

Two continuous control pedals (or one pedal and one breath controller)

Four foot switches



Solo Button

Mutes all zones in setup except the current one. The button of the zone being soloed glows red.

Mixdown Button

Brings up the Mixdown page, as shown in the following diagram. From this page you can choose how the K2600's physical sliders function during MIDI mixdown. In the example below, Sliders A-H will control the volume level of MIDI channels 1-8. By pressing the **Pan** soft button, you would change the function of the sliders to control panning for channels 1-8; or, you could press the **9-16** soft button to have the sliders affect channels 9-16.

You can also use the cursor buttons to highlight the pan or volume control for a channel and use the Alpha Wheel or **Plus/Minus** buttons to change the pan or volume level. In the screen below, for example, you could use the Alpha Wheel to control panning on channel 9 at the same time that you are using the sliders to control volume on channels 1-8.

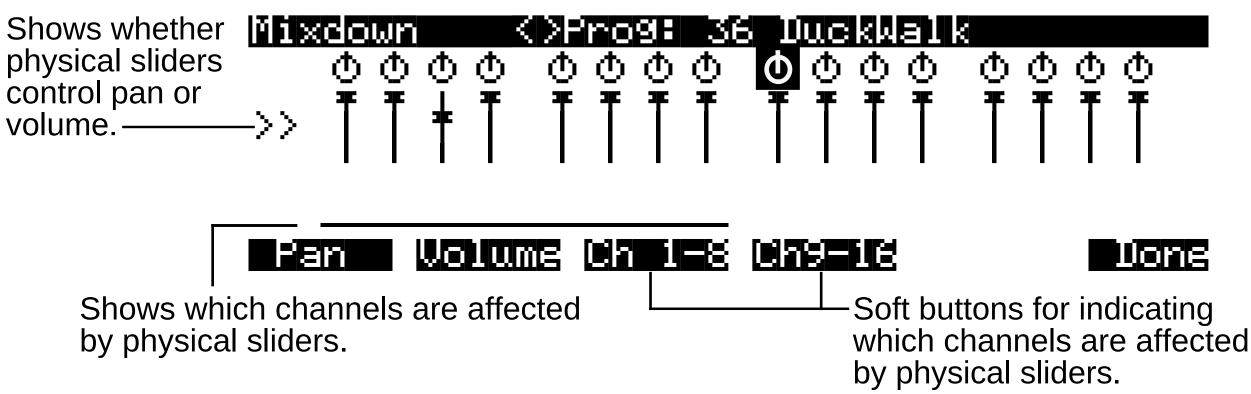
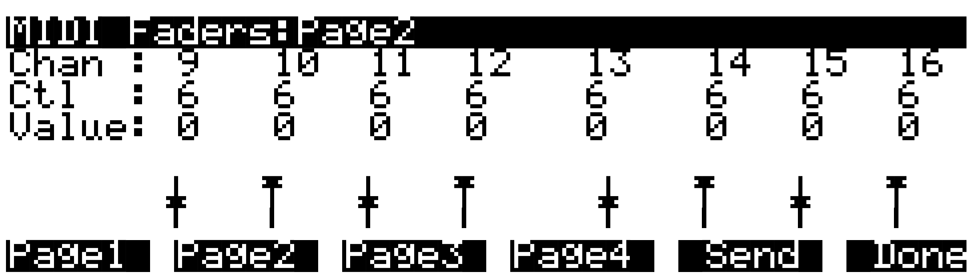


Figure 1-1 Mixdown Control

MIDI Faders button

When you press the **MIDI Faders** button, the K2600's sliders take on the functions assigned on the current MIDI Faders page. From the MIDI Faders display you can define four different pages that define how the K2600's physical sliders will work. In the display shown below, for example, the eight sliders are each defined to send MIDI 6 (Data) on Channels 9 through 16. Press one of the **Page** soft buttons to use (or create) a different page of MIDI fader assignments. Use the **Send** soft button to transmit values without moving the faders.

The MIDI Faders pages is saved as part of the Master table object.



Assignable Controllers (Buttons 1–8 and Sliders A–H)

The function of these controllers will depend on how they've been defined within a setup. Buttons 1–8 control either zone muting or KB3 features, depending on the value of the value of the Mutes parameter on the COMMON page in the Setup Editor. The SLIDER and SLID/2 pages configure the functions of Sliders A–H.

PSw1, PSw2 (Buttons 9 and 10)

The function of these controllers depends on how they've been defined on the SWITCH page in the Setup Editor.

Record, Play/Pause, Stop

These buttons duplicate the functions of the corresponding soft buttons in Song mode, allowing you to conveniently record, play, pause, and stop the current song.

Special Button Functions

The Mode buttons and the **Chan/Bank Down** button have additional functions, depending on the mode or editor you're in. When you're in the Program or Setup Editor, they function according to the blue labeling under each button. They also work as track mutes on the MIX page of Song mode.

When you're in the Sample Editor, the **Program**, **Setup**, **Q Access**, **MIDI**, **Master**, and **Song** mode buttons function according to the orange labeling near each button. Table 1-2 describes all of the special button functions. This table also appears as Table 5-1 on page 5-8 of the *Musician's Guide*.

Button	Mode or Editor			
White Blue Orange	Program Editor (Blue)	Setup Editor (Blue)	Song Mode	Sample Editor (Orange)
Program Mute 1 Zoom-	Mutes Layer 1 of current program, or mutes current layer of current drum program	Mutes Zone 1 of current setup if 3 or fewer zones; mutes current zone of current setup if more than 3 zones	On MIX page, mutes Track 1 or 9	On TRIM and LOOP pages, decreases horizontal dimension of current sample in display
Setup Mute 2 Zoom+	Mutes Layer 2 of current program, or solos current layer of current drum program	Mutes Zone 2 of current setup if 3 or fewer zones; solos current zone of current setup if more than 3 zones	On MIX page, mutes Track 2 or 10	On TRIM and LOOP pages, increases horizontal dimension of current sample in display
Q Access Mute 3 Samp / Sec	Mutes Layer 3 of current program, or solos current layer of current drum program	Mutes Zone 3 of current setup if 3 or fewer zones; solos current zone of current setup if more than 3 zones	On MIX page, mutes Track 3 or 11	Toggles between units used to identify location within sample— either number of samples from start, or time in seconds from start
Effects FX Bypass	Bypasses (mutes) current program's FX preset (plays program dry)	Bypasses (mutes) current setup's studio (plays studio dry)	On MIX page, mutes Track 4 or 12	

Table 1-2 Special Button Functions

Button	Mode or Editor			
White Blue Orange	**Program Editor (Blue)**	**Setup Editor (Blue)**	**Song Mode**	**Sample Editor (Orange)**
MIDI Previous Pg Gain -	Successive presses take you back to four most recent editor pages; 5th press takes you to ALG page	Successive presses take you back to four most recent editor pages; 5th press takes you to CH/PRG page	On MIX page, mutes Track 5 or 13	On TRIM and LOOP pages, decreases vertical dimension of current sample in display
Master Mark Gain +	“Remembers” current editor page, so you can recall multiple pages with Jump button; asterisk appears before page name to indicate that it’s marked; unmark pages by pressing Mark when page is visible	Same as for Program Editor; pages common to both editors are marked or unmarked for *both* editors	On MIX page, mutes Track 6 or 14	On TRIM and LOOP pages, increases vertical dimension of current sample in display
Song Jump Link	Jumps to marked pages in order they were marked	Jumps to marked pages in order they were marked	On MIX page, mutes Track 7 or 15	Preserves interval between Start, Alt, Loop, and End points of current sample; press again to unlink
Disk Compare	Negates effect of unsaved edits and plays last-saved (unedited) version of object being edited	Same as for Program mode; display reminds you that you’re comparing; press any button to return to edited version	On MIX page, mutes Track 8 or 16	
Chan / Bank Layer / Zone	In Program Editor, these two buttons scroll through layers of current program; in Effects Editor, scroll through FX presets; in Keymap Editor, scroll through velocity levels of current keymap; in Setup Editor, scroll through zones of current setup; in Quick Access mode, scroll through entries in current Quick Access bank		Change recording track	
Edit	Whenever cursor is highlighting an editable object or parameter, takes you to corresponding editor or programming page			

Table 1-2 Special Button Functions (Continued)

Special Button Functions: Double Button Presses

Pressing two or more related buttons simultaneously executes a number of special functions depending on the currently selected mode. Make sure to press them at exactly the same time. The following table also appears as Table 3-1 on page 3-6 of the *Musician's Guide*.

In this mode or editor…	…pressing these buttons simultaneously…	…does this:
Program mode	Octav-, Octav+	Reset MIDI transposition to 0 semitones. Double-press again to go to previous transposition.
	Chan–, Chan+	Set current MIDI channel to 1.
	Plus/Minus	Step to next Program bank (100, 200, etc.)
Master mode	Chan/Bank	Enables Guitar/Wind Controller mode.
Song mode	Left/Right cursor buttons	Toggle between Play and Stop.
	Up/Down cursor buttons	Toggle between Play and Pause.
	Chan/Bank	Select all tracks on any TRACK page in Song Editor.
Disk mode	2 leftmost soft buttons	Issue SCSI Eject command to currently selected SCSI device.
	Chan/Bank	Hard format SCSI device. List selected objects when saving objects.
	Left/Right cursor buttons	Select all items in a list. Move cursor to end of name in naming dialog.
	up/down cursor buttons	Clear all selections in a list. Move cursor to beginning of name in naming dialog.
Program Editor	Chan/Bank	Select Layer 1.
Keymap Editor	Plus/Minus	With cursor on the Coarse Tune parameter, toggles between default Coarse Tune of sample root and transposition of sample root.
Sample Editor	2 leftmost soft buttons	Toggle between default zoom setting and current zoom setting.
	Plus/Minus buttons	Set the value of the currently selected parameter at the next *zero crossing.*
Any Editor	Plus/Minus	Scroll through the currently selected parameter's list of values in regular or logical increments (varies with each parameter).
	2 leftmost soft buttons	Reset MIDI transposition to 0 semitones. Double-press again to go to previous transposition.
	Center soft buttons	Select Utilities menu (MIDIScope, Stealer, etc.).
	2 rightmost soft buttons	Sends all notes/controllers off message on all 16 channels (same as Panic soft button).
	Left/Right cursor buttons	Toggle between Play and Stop of current song.
	Up/Down cursor buttons	Toggle between Play and Pause of current song.
Save Dialog	Plus/Minus buttons	Toggle between next free ID and original ID.

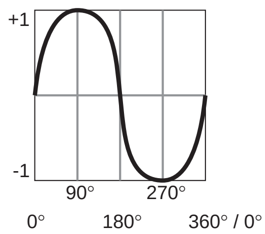
Table 1-3 Double Button Presses

Chapter 2
LFOs

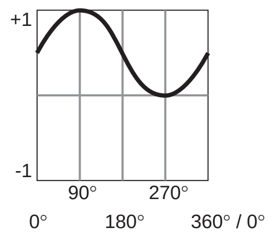
LFO Shapes

LFO Shape	Displayed As
Sine	Sine
Positive Sine	+Sine
Square	Square
Positive Square	+Squar
Triangle	Triang
Positive Triangle	+Trian
Rising Sawtooth	Rise S
Positive Rising Sawtooth	+Rise
Falling Sawtooth	Fall S
Positive Falling Sawtooth	+Fall
3 Step	3 Step
Positive 3 Step	+3 Ste
4 Step	4 Step
Positive 4 step	+4 Ste
5 Step	5 Step
Positive 5 Step	+5 Ste
6 Step	6 Step
Positive 6 Step	+6 Ste
7 Step	7 Step
Positive 7 Step	+7 Ste
8 Step	8 Step
Positive 8 Step	+8 Ste
10 Step	10 Ste
Positive 10 Step	+10 St
12 Step	12 Ste
Positive 12 Step	+12 St

Sine



Positive Sine



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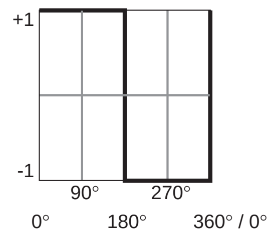
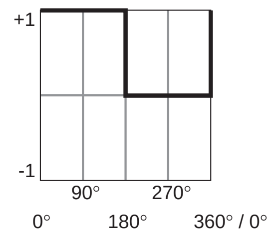
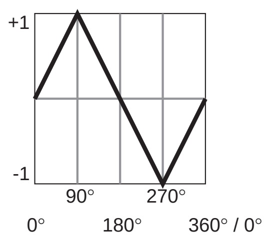


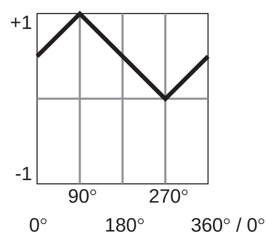
Positive Sq uare



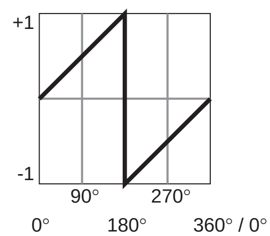
Triangle



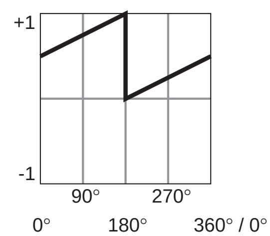
Positive Triangle



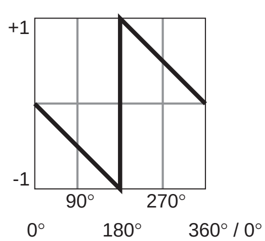
Rising Sawtooth



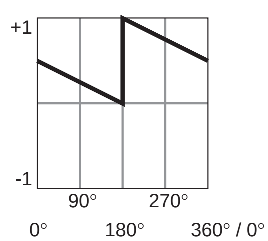
Positive Rising Sawtooth



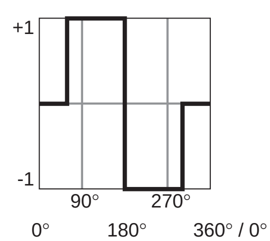
Falling Sawtooth



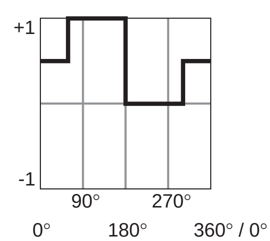
Positive Falling Sawtooth



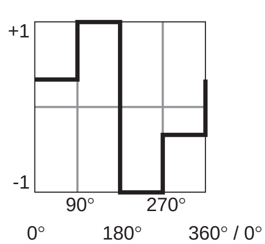
3 Step



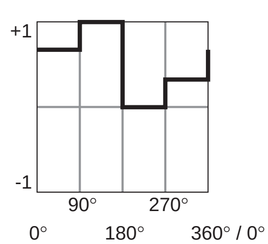
Positive 3 Step



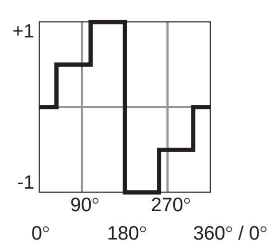
4 Step



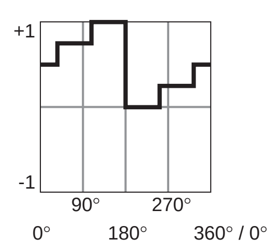
Positive 4 Step



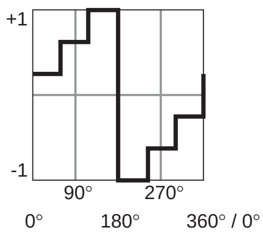
5 Step



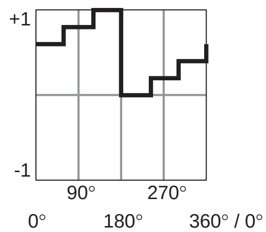
Positive 5 Step



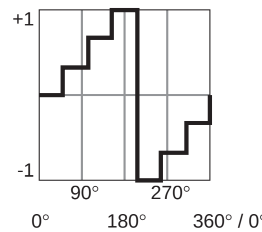
6 Step



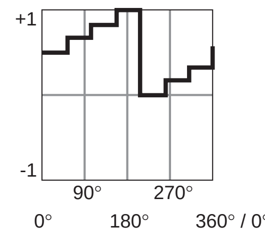
Positive 6 Step



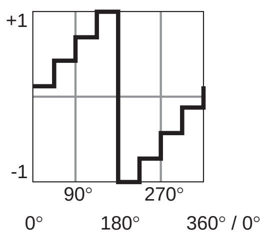
7 Step



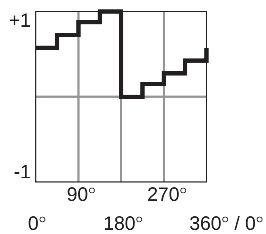
Positive 7 Step



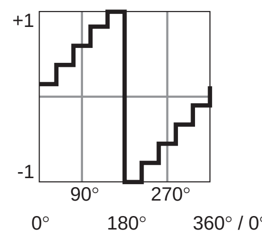
8 Step



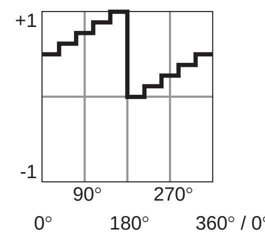
Positive 8 Step



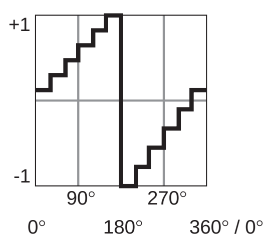
10 Step



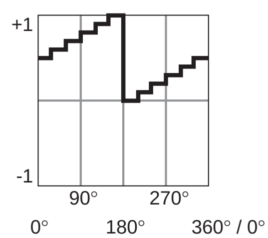
Positive 10 Step



12 Step

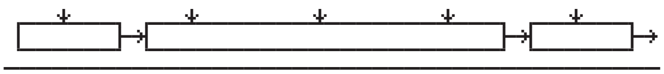


Positive 12 Step



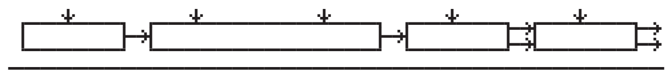
Chapter 3
DSP Algorithms

Algorithm 1



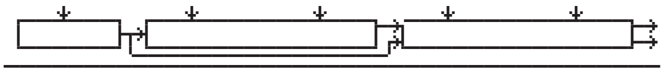
PITCH	HIFREQ STIMULATOR	AMP
	PARAMETRIC EQ	
	STEEP RESONANT BASS	
	4POLE LOPASS W/SEP	
	4POLE HIPASS W/SEP	
	TWIN PEAKS BANDPASS	
	DOUBLE NOTCH W/SEP	
	NONE	

Algorithm 2



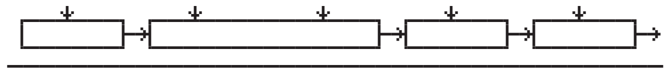
PITCH	2PARAM SHAPER	PANNER	AMP
	2POLE LOWPASS		
	BANDPASS FILT		
	NOTCH FILTER		
	2POLE ALLPASS		
	PARA BASS		
	PARA TREBLE		
	PARA MID		
	NONE		

Algorithm 3



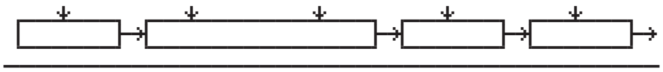
PITCH	2PARAM SHAPER	AMP U AMP L
	2POLE LOWPASS	BAL AMP
	BANDPASS FILT	
	NOTCH FILTER	
	2POLE ALLPASS	
	NONE	

Algorithm 4



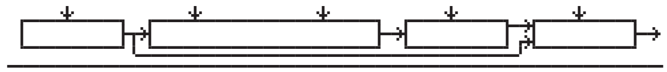
PITCH	2PARAM SHAPER	LPCLIP	AMP
	2POLE LOWPASS	SINE+	
	BANDPASS FILT	NOISE+	
	NOTCH FILTER	LOPASS	
	2POLE ALLPASS	HIPASS	
	PARA BASS	ALPASS	
	PARA TREBLE	GAIN	
	PARA MID	SHAPER	
	NONE	DIST	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 5



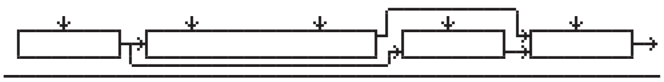
PITCH	2PARAM SHAPER	LP2RES	AMP
	2POLE LOWPASS	SHAPE2	
	BANDPASS FILT	BAND2	
	NOTCH FILTER	NOTCH2	
	2POLE ALLPASS	LOPAS2	
	PARA BASS	HIPAS2	
	PARA TREBLE	LPGATE	
	PARA MID	NONE	
	NONE		

Algorithm 6



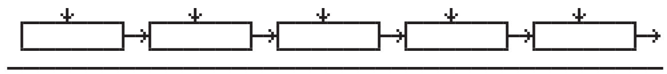
PITCH	2PARAM SHAPER	LPCLIP	x AMP
	2POLE LOWPASS	SINE+	+ AMP
	BANDPASS FILT	NOISE+	! AMP
	NOTCH FILTER	LOPASS	
	2POLE ALLPASS	HIPASS	
	NONE	ALPASS	
		GAIN	
		SHAPER	
		DIST	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 7



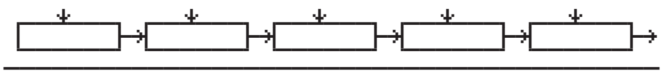
PITCH	2PARAM SHAPER	LPCLIP	x AMP
	2POLE LOWPASS	SINE+	+ AMP
	BANDPASS FILT	NOISE+	! AMP
	NOTCH FILTER	LOPASS	
	2POLE ALLPASS	HIPASS	
	NONE	ALPASS	
		GAIN	
		SHAPER	
		DIST	
		SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 8



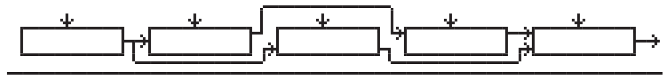
PITCH	LOPASS	LOPASS	LPCLIP	AMP
	HIPASS	HIPASS	SINE+	
	ALPASS	ALPASS	NOISE+	
	GAIN	GAIN	LOPASS	
	SHAPER	SHAPER	HIPASS	
	DIST	DIST	ALPASS	
	PWM	SW+SHP	GAIN	
	SINE	SAW+	SHAPER	
	LF SIN	WRAP	DIST	
	SW+SHP	NONE	SW+SHP	
	SAW+		SAW+	
	SAW		SW+DST	
	LF SAW		NONE	
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 9



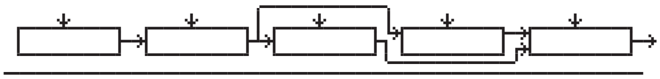
PITCH	LOPASS	LOPASS	LP2RES	AMP
	HIPASS	HIPASS	SHAPE2	
	ALPASS	ALPASS	BAND2	
	GAIN	GAIN	NOTCH2	
	SHAPER	SHAPER	LOPAS2	
	DIST	DIST	HIPAS2	
	PWM	SW+SHP	LPGATE	
	SINE	SAW+	NONE	
	LF SIN	WRAP		
	SW+SHP	NONE		
	SAW+			
	SAW			
	LF SAW			
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 10



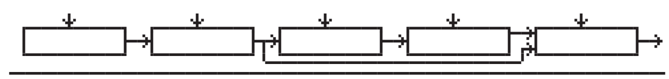
PITCH	LOPASS	LOPASS	LPCLIP	x AMP
	HIPASS	HIPASS	SINE+	+ AMP
	ALPASS	ALPASS	NOISE+	! AMP
	GAIN	GAIN	LOPASS	
	SHAPER	SHAPER	HIPASS	
	DIST	DIST	ALPASS	
	PWM	SINE	GAIN	
	SINE	LF SIN	SHAPER	
	LF SIN	SW+SHP	DIST	
	SW+SHP	SAW+	SW+SHP	
	SAW+	SAW	SAW+	
	SAW	LF SAW	SW+DST	
	LF SAW	SQUARE	NONE	
	SQUARE	LF SQR		
	LF SQR	WRAP		
	WRAP	NONE		
	NONE			

Algorithm 11



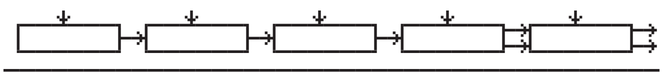
PITCH	LOPASS	LOPASS	LPCLIP	x AMP
	HIPASS	HIPASS	SINE+	+ AMP
	ALPASS	ALPASS	NOISE+	! AMP
	GAIN	GAIN	LOPASS	
	SHAPER	SHAPER	HIPASS	
	DIST	DIST	ALPASS	
	PWM	SINE	GAIN	
	SINE	LF SIN	SHAPER	
	LF SIN	SW+SHP	DIST	
	SW+SHP	SAW+	SINE	
	SAW+	SAW	LF SIN	
	SAW	LF SAW	SW+SHP	
	LF SAW	SQUARE	SAW+	
	SQUARE	LF SQR	SW+DST	
	LF SQR	WRAP	NONE	
	WRAP	NONE		
	NONE			

Algorithm 12



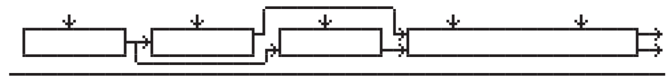
PITCH	LOPASS	LOPASS	LPCLIP	x AMP
	HIPASS	HIPASS	SINE+	+ AMP
	ALPASS	ALPASS	NOISE+	! AMP
	GAIN	GAIN	LOPASS	
	SHAPER	SHAPER	HIPASS	
	DIST	DIST	ALPASS	
	PWM	PWM	GAIN	
	SINE	SINE	SHAPER	
	LF SIN	LF SIN	DIST	
	SW+SHP	SW+SHP	SW+SHP	
	SAW+	SAW+	SAW+	
	SAW	SAW	SW+DST	
	LF SAW	LF SAW	NONE	
	SQUARE	SQUARE		
	LF SQR	LF SQR		
	WRAP	WRAP		
	NONE	NONE		

Algorithm 13



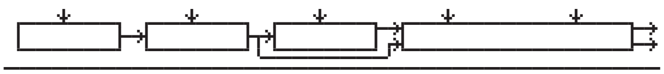
PITCH	LOPASS	LOPASS	PANNER	AMP
	HIPASS	HIPASS		
	ALPASS	ALPASS		
	GAIN	GAIN		
	SHAPER	SHAPER		
	DIST	DIST		
	PWM	SW+SHP		
	SINE	SAW+		
	LF SIN	WRAP		
	SW+SHP	NONE		
	SAW+			
	SAW			
	LF SAW			
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 14



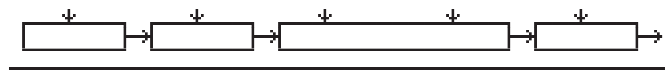
PITCH	LOPASS	LOPASS	AMP U AMP L
	HIPASS	HIPASS	BAL AMP
	ALPASS	ALPASS	
	GAIN	GAIN	
	SHAPER	SHAPER	
	DIST	DIST	
	SINE	SINE	
	LF SIN	LF SIN	
	SW+SHP	SW+SHP	
	SAW+	SAW+	
	SAW	SAW	
	LF SAW	LF SAW	
	SQUARE	SQUARE	
	LF SQR	LF SQR	
	WRAP	WRAP	
	NONE	NONE	

Algorithm 15



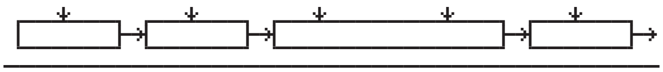
PITCH	LOPASS	LOPASS	AMP U AMP L
	HIPASS	HIPASS	BAL AMP
	ALPASS	ALPASS	
	GAIN	GAIN	
	SHAPER	SHAPER	
	DIST	DIST	
	PWM	SINE	
	SINE	LF SIN	
	LF SIN	SW+SHP	
	SW+SHP	SAW+	
	SAW+	SAW	
	SAW	LF SAW	
	LF SAW	SQUARE	
	SQUARE	LF SQR	
	LF SQR	WRAP	
	WRAP	NONE	
	NONE		

Algorithm 16



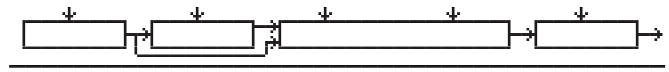
PITCH	LOPASS	PARA BASS	AMP
	HIPASS	PARA TREBLE	
	ALPASS	NONE	
	GAIN		
	SHAPER		
	DIST		
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		
	NONE		

Algorithm 17



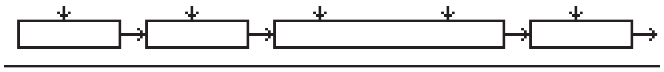
PITCH	LOPASS	SHAPE MOD OSC	AMP
	HIPASS	AMP MOD OSC	
	ALPASS	NONE	
	GAIN		
	SHAPER		
	DIST		
	PWM		
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		
	NONE		

Algorithm 18



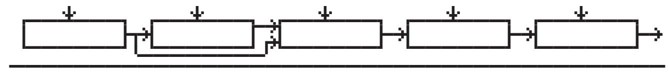
PITCH	LOPASS	x SHAPEMOD OSC	AMP
	HIPASS	+ SHAPEMOD OSC	
	ALPASS	NONE	
	GAIN		
	SHAPER		
	DIST		
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		
	NONE		

Algorithm 19



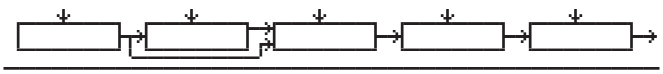
PITCH	LOPAS2	SHAPE MOD OSC	AMP
	NONE	NONE	

Algorithm 20



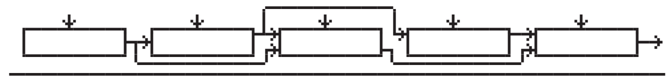
PITCH	LOPASS	x GAIN	LPCLIP	AMP
	HIPASS	+ GAIN	SINE+	
	ALPASS	XFADE	NOISE+	
	GAIN	AMPMOD	LOPASS	
	SHAPER	NONE	HIPASS	
	DIST		ALPASS	
	SINE		GAIN	
	LF SIN		SHAPER	
	SW+SHP		DIST	
	SAW+		SW+SHP	
	SAW		SAW+	
	LF SAW		SW+DST	
	SQUARE		NONE	
	LF SQR			
	WRAP			
	NONE			

Algorithm 21



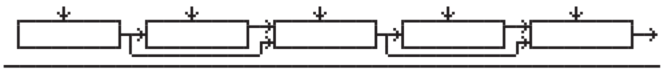
PITCH	LOPASS	x GAIN	LP2RES	AMP
	HIPASS	+ GAIN	SHAPE2	
	ALPASS	XFADE	BAND2	
	GAIN	AMPMOD	NOTCH2	
	SHAPER	NONE	LOPAS2	
	DIST		HIPAS2	
	SINE		LPGATE	
	LF SIN		NONE	
	SW+SHP			
	SAW+			
	SAW			
	LF SAW			
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 22



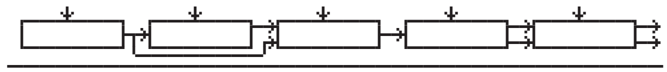
PITCH	LOPASS	x GAIN	LPCLIP	x AMP
	HIPASS	+ GAIN	SINE+	+ AMP
	ALPASS	XFADE	NOISE+	! AMP
	GAIN	AMPMOD	LOPASS	
	SHAPER	NONE	HIPASS	
	DIST		ALPASS	
	SINE		GAIN	
	LF SIN		SHAPER	
	SW+SHP		DIST	
	SAW+		SINE	
	SAW		LF SIN	
	LF SAW		SW+SHP	
	SQUARE		SAW+	
	LF SQR		SW+DST	
	WRAP		NONE	
	NONE			

Algorithm 23



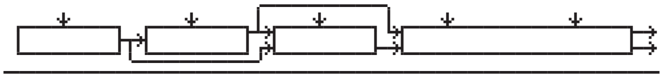
PITCH	LOPASS	x GAIN	LPCLIP	x AMP
	HIPASS	+ GAIN	SINE+	+ AMP
	ALPASS	XFADE	NOISE+	! AMP
	GAIN	AMPMOD	LOPASS	
	SHAPER	NONE	HIPASS	
	DIST		ALPASS	
	SINE		GAIN	
	LF SIN		SHAPER	
	SW+SHP		DIST	
	SAW+		SINE	
	SAW		LF SIN	
	LF SAW		SW+SHP	
	SQUARE		SAW+	
	LF SQR		SW+DST	
	WRAP		NONE	
	NONE			

Algorithm 24



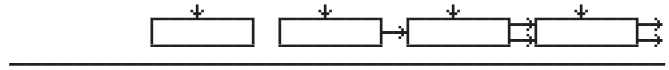
PITCH	LOPASS	x GAIN	PANNER	AMP
	HIPASS	+ GAIN		
	ALPASS	XFADE		
	GAIN	AMPMOD		
	SHAPER	NONE		
	DIST			
	SINE			
	LF SIN			
	SW+SHP			
	SAW+			
	SAW			
	LF SAW			
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 25



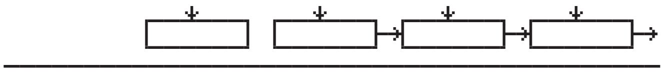
PITCH	LOPASS	x GAIN	AMP U AMP L
	HIPASS	+ GAIN	BAL AMP
	ALPASS	XFADE	
	GAIN	AMPMOD	
	SHAPER	NONE	
	DIST		
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		
	NONE		

Algorithm 26



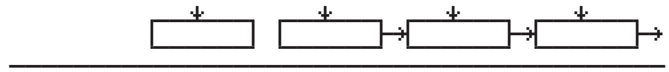
SYNC M	SYNC S	PANNER	AMP

Algorithm 27



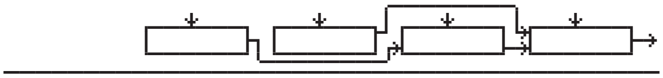
SYNC M	SYNC S	LPCLIP	AMP
		SINE+	
		NOISE+	
		LOPASS	
		HIPASS	
		ALPASS	
		GAIN	
		SHAPER	
		DIST	
		SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 28



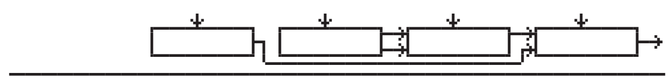
SYNC M	SYNC S	LP2RES	AMP
		SHAPE2	
		BAND2	
		NOTCH2	
		LOPAS2	
		HIPAS2	
		LPGATE	
		NONE	

Algorithm 29



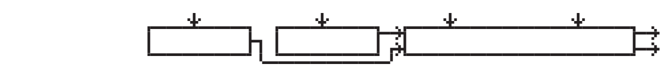
SYNC M	SYNC S	LPCLIP	x AMP
		SINE+	+ AMP
		NOISE+	! AMP
		LOPASS	
		HIPASS	
		ALPASS	
		GAIN	
		SHAPER	
		DIST	
		SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 30



SYNC M	SYNC S	LPCLIP	x AMP
		SINE+	+ AMP
		NOISE+	! AMP
		LOPASS	
		HIPASS	
		ALPASS	
		GAIN	
		SHAPER	
		DIST	
		SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 31



SYNC M	SYNC S	AMP U AMP L
		BAL AMP

Chapter 4
Control Sources

Control sources are assigned as values for control source parameters, like Src1 and Src2, Depth Control for Src2, and LFO rate control. Assigning a control source to one of these parameters is like connecting control source outputs to various inputs on early modular synthesizers. You can think of each control source parameter as the input to a synthesizer module, and the values for those parameters as the outputs of modules generating control signals.

For the control sources to have an effect, two things have to happen. First, the control source must be assigned as the value for (patched to) a control source parameter like Src1. In other words, for a control source parameter to have an effect, it must be programmed to respond to a particular control message. Second, the control source must generate a signal. The level of the control source's signal determines how much effect it has on the control source parameter to which it's assigned.

In terms of generating signals, there are two types of control sources. The first, which might be called hardware control sources, require some physical movement to transmit them. The control source called MWheel (MIDI 01) is probably the most prominent example of this type of control source. When you move your MIDI controller's Mod Wheel, it sends a Modulation message (MIDI 01), unless you've programmed it to send something else. By default, when the K2600 receives a MIDI 01 message, it responds by sending a control signal to whatever control source is assigned as the value for the MWhl parameter on the MIDI-mode RECEIVE page. Of course, you can program the MWhl parameter to send any available control source signal in response to MIDI 01 messages.

Some of these hardware control sources have physical controls "hard-wired" to transmit them. That is, there are certain physical controls that *always* generate these control signals. Every time you strike one of your MIDI source's keys (or pluck a string, or whatever), for example, a Note On message is generated, along with an Attack Velocity message. So any time you strike a key, any control source parameter that has AttVel assigned as its value will be affected by the Attack Velocity message. Similarly, every time you move the physical Pitch Wheel, a PWheel message is generated. Whether this affects anything depends on whether you have assigned any control source parameters to respond to the PWheel message (in other words, whether any control source parameter has PWheel assigned as its value).

In the Setup Editor you'll find several parameters that correspond to the standard physical controllers found on many keyboards. These parameters and their default values are listed in Table C-1 on page C-2. The values you assign for these parameters determine which control messages will be transmitted to the K2600 and to its MIDI Out port when you move the corresponding controls on your MIDI source. If you look at the WHEEL page in the Setup Editor, you'll see that the parameter called MWhl has a default value of **MWheel**. You can interpret this as follows: "Moving the Mod Wheel on my MIDI source sends the MWheel (Modulation, MIDI 01) message to the K2600's sound engine, and, if the K2600's LocalKbdCh parameter matches my controller's transmit channel, also sends it to the K2600's MIDI Out port."

If you change the value of the MWhl parameter, the Mod Wheel will no longer send the MWheel message, and any control source parameter with **MWheel** assigned as its value will no longer respond to movement of the Mod Wheel. All of the control assignment parameters in the Setup Editor can be programmed to send any of the MIDI controller numbers. For example, if you assign **Foot** (MIDI 04) as the value for the Press parameter, then generating mono pressure messages from your MIDI source will send a Foot (MIDI 04) message to the K2600's sound engine, and will affect any control source parameter that has **Foot** assigned as its value. If the value for the K2600's LocalKbdCh parameter matches your MIDI controller's transmit channel, then in this case the Foot message will be sent to the K2600's MIDI Out port as well, when you generate mono pressure messages from your MIDI controller.

The other type of control source is independent of the movement of physical controls. These control sources generate their control signals internally, and might be called software control sources. They either run automatically (like A Clock and RandV1), or they're programmed to generate their signals according to parameters of their own (as with the LFOs and FUNs). The software control sources must have some nonzero value set for one or more of their parameters before they'll generate control signals.

To summarize, there are two different cases in which you'll assign control sources. One, the transmit case, determines what control message will be sent by a particular physical control. For example, MWheel is set by default to be transmitted by the Mod Wheel. The other case, the receive case, determines which control message will activate a particular control source parameter. For example, if you assign **MPress** as the value for the Src1 parameter on the PITCH page in the Program Editor, then that layer's pitch will be affected whenever an MPress message is generated by any physical controller.

Control Source Lists

There's one long list of control sources stored in the K2600's memory, although not all control sources are available for all control source parameters. With time you'll become familiar with the types of control sources available for various control source parameters.

The available list of control sources varies depending on the type of control source parameter you're programming. There are four basic types: MIDI control sources, local control sources, global control sources, and FUNs.

When you're setting the control assignment parameters in the Setup Editor, you'll see only the portion of the Control Source list that has values appropriate to MIDI controller messages. Consequently we refer to this subset of the Main Control Source list as the MIDI Control Source list.

You'll see variations on the Main Control Source list as you program the other control source parameters. We'll explain these variations, but it's not important that you memorize each variation. The lists differ to prevent you from assigning a control source where it would be ineffective. All you have to do is to scroll through the list of control sources available for any given control source parameter, and choose from the available values.

If you're programming one of the FUNs, you'll see the Main Control Source list, which includes almost every control source from the MIDI Control Source list (with the exception of Data Inc, Data Dec, and Panic, which belong exclusively to the MIDI Control Source list). The list for the FUNs also includes a set of constant values, that set an unvarying control signal level for one or both of the FUN's inputs.

For most other control source parameters, you'll see the Main Control Source list (without the FUN constants and the three special MIDI control sources we mentioned above). There are two exceptions to this rule, which have to do with global control source parameters. Globals affect every note in each program's layer(s). Consequently they can't use local control sources as their values, since local control sources affect each note independently.

One control source parameters is always global: the Enable parameter on the LAYER page (Program Editor). When programming this parameter, you'll see the Main Control Source list minus the three special MIDI control sources, minus the following local control sources:

Note St	VTRIG2
Key St	RandV1
KeyNum	RandV2
BKeyNum	ASR1
AttVel	LFO1
InvAVel	FUN1
PPress	FUN3
BPPress	Loop St
RelVel	PB Rate
Bi-AVel	AtkSt
VTRIG1	Rel St

Finally, if you've turned on the Globals parameter on the COMMON page in the Program Editor, the available values for GLFO2, and the values for GASR2's trigger will lack the local control sources listed above, as well as the three special MIDI control sources and the FUN constants. The available values for GFUN2 and GFUN4 will exclude the same list of local control sources, but will include the FUN constants.

Descriptions of Control Sources

This section is organized into two sets of descriptions: the MIDI Control Source list, and the rest of the control sources. The numeral preceding the name of each control source can be entered on the alphanumeric pad to select the control source directly (press **Enter** after typing the numeral).

Many of the MIDI control sources are assigned as default values for the control assignment parameters in the Setup Editor. We'll indicate these assignments as they appear, simply by mentioning that they're the default control source for a control assignment parameter.

MIDI Control Source List

With a few exceptions, the MIDI control sources correspond to the standard MIDI controller numbers used by every MIDI device.

128 OFF

This value eliminates the effect of any control source parameter to which it's assigned.

0, 33 Mono Pressure (MPress)

Many of the K2600's factory programs are assigned to modify parameters such as pitch, filter cutoff frequency, and depth control when MPress messages are received. The mono pressure (Press) control assignment parameters in MIDI and Setup modes are set by default to transmit MPress messages when mono pressure messages are received from a controller.

1 MIDI 01 (MWheel)

Many factory programs are assigned to respond to MWheel messages. The MWhl parameter in the Setup Editor is set by default to transmit MWheel.

2 MIDI 02 (Breath)

3 MIDI 03

4 MIDI 04 (Foot)

This is the standard MIDI Controller number for continuous control foot pedals. It's the default value for the CPedal control assignment parameter, so a control pedal on your MIDI controller which sends MIDI controller 04 messages will send MIDI controller 04 messages to the K2600 by default.

5 MIDI 05 (PortTim)

This is the standard MIDI controller number for portamento time control. The K2600 always responds to this control message. For any program that has portamento turned on (on the COMMON page in the Program Editor), MIDI Portamento Time messages received via MIDI will affect the rate of the program's portamento.

6 MIDI 06 (Data)

MIDI 06 is the standard MIDI controller number for data entry. The Slider A parameter on the SLIDER page in the Setup Editor is set by default to transmit this message, and can be used to select programs and edit parameters on MIDI slaves if your controller can send it.

7 MIDI 07 (Volume)

This is the standard MIDI controller number for volume. The Volume parameter on the CHANNELS page in MIDI mode will respond to MIDI controller 07 unless the VolLock parameter is turned on.

8 MIDI 08 (Balance)

9 MIDI 09

10 MIDI 10 (Pan)

MIDI controller 10 is defined as Pan control. The Pan parameter on the CHANNELS page in MIDI mode will respond to MIDI controller 10 unless the PanLock parameter is turned on.

11 MIDI 11 (Express)

12—14 MIDI 12—14

15 MIDI 15 (AuxBend2)

The K2600 interprets MIDI Controller 15 as AuxBend2, which is assigned by default to the short ribbon (below the pitch and mod wheels) on keyboard models of the instrument. A value of 64 is centered.

16—19 MIDI 16—19 (Ctl A—D)

20 MIDI 20

21 MIDI 21 (AuxBend1)

The K2600 interprets MIDI Controller 21 as AuxBend1, which is assigned by default to the long ribbon (above the keyboard) on keyboard models of the instrument. A value of 64 is centered.

22—31 MIDI 22—31

64 MIDI 64 (Sustain)

This is the standard MIDI Controller number for Sustain. The control assignment parameter FtSw1 is set by default to MIDI Controller 64, so a switch pedal on your MIDI controller that sends MIDI 64 will send sustain messages to the K2600 by default. The K2600 will always respond to sustain messages by sustaining currently active notes.

65 MIDI 65 (PortSw)

This is the standard MIDI Controller number for Portamento Switch. The Portamento parameter on the COMMON page in the Program Editor always responds to this controller, and will turn Portamento on for monophonic programs when the controller signal is at 64 or above. It won’t affect polyphonic programs.

66 MIDI 66 (SostPD)

MIDI Controller 66 is defined as Sostenuto Switch. The control assignment parameter FtSw2 is set by default to MIDI Controller 66, so a switch pedal on your MIDI controller that sends MIDI 66 will send sostenuto messages to the K2600 by default. The K2600 will always respond to sostenuto messages.

67 MIDI 67 (SoftPd)

This is the standard MIDI Controller number for Soft Pedal. The K2600 will always respond to Soft pedal messages.

68 MIDI 68

69 MIDI 69 (FrezPd)

The K2600 will always respond to this message. It causes all notes to be frozen at their current amplitude levels while the function is on.

70—74 MIDI 70—74

75 MIDI 75 (LegatoSw)

The K2600 always responds to this message. When a MIDI Controller 75 message with a value above 64 is received, the K2600 will force polyphonic programs to be monophonic.

76—79 MIDI 76—79

80—83 MIDI 80—83 (Ctl E—H)

84—90 MIDI 84—90

91 MIDI 91 (FXDep)

The MIDI specification defines this Controller as External Effects Depth. If the FX Mode parameter is set to **Master**, and the FX Channel parameter is set to a specific MIDI channel, the K2600 will respond to this message when it is received on the FX channel. It responds by adjusting the Wet/Dry mix of the current studio.

92—95 MIDI 92—95

96 MIDI 96 (DataInc)

This is defined as Data Increment. It's intended to be assigned to a switch control. When the control is on (value 127), the currently selected parameter's value will be increased by one increment. This could be assigned to FtSw2, for example, to scroll through the program list while in Program mode.

97 MIDI 97 (DataDec)

This is defined as Data Decrement. It's intended to be assigned to a switch control. When the control is on (value 127), the currently selected parameter's value will be decreased by one increment.

123 MIDI 123 (Panic)

The K2600 always responds to this message by sending an All Notes Off and All Controllers Off message on all 16 MIDI channels.

Main Control Source List

This list contains all but the last three control sources in the MIDI Control Source list. It also contains the following control sources. All are local unless specified as global.

32 Channel State (Chan St)

Chan St refers to whether any notes are currently active on a given MIDI channel. Chan St switches on whenever a note is started, and switches off when a Note Off has been received for each current note on that channel, even if notes are sustained.

33 Mono Pressure (MPress)

This is the same as the MPress control source in the MIDI Control Source list, but is assigned by entering 33 on the alphanumeric pad when used with a parameter that takes its values from the Main Control Source list.

34 Bipolar Mono Pressure (BMPress)

This control source generates a control signal of -1 when the value of the control to which it's assigned is at its minimum, and +1 when the control is at its maximum. For example, if you had the MPress control assignment parameter assigned to send BMPress, and you had Src1 on a program layer's PITCH page assigned to **BMPress**, with its depth parameter set to **1200 cents**, then the layer would be transposed down an octave when no pressure (value 0) was applied to your controller's keys (assuming it sends mono pressure). Maximum pressure (value 127) would transpose the layer up an octave, while a pressure level of 64 would leave the pitch unchanged.

35 Pitch Wheel Message (PWheel)

The K2600 is hard-wired to respond to this message. Any parameter with **PWheel** assigned as its value will be affected when your MIDI controller's Pitch Wheel is moved.

36 Bipolar Mod Wheel (Bi-Mwl)

This control source will always respond to MIDI controller 01 (MWheel). Control source parameters set to this value will generate control signals of -1 when the MIDI Controller 01 message value is 0, and will generate a control signal of +1 when the MIDI Controller 01 message is at 127, scaling all values in between. For example, you might set Src1 on a program layer's PITCH page to a value of **Bi-Mwl**, and its depth parameter to **1200 cents**. Then as long as the MWhl control assignment parameter is set to a value of **MWheel**, your controller's Mod Wheel will be bipolar; in this case it will bend the layer's pitch down as you move the Mod Wheel toward minimum, and bend the pitch up as you move the Mod Wheel toward maximum.

37 Absolute Value of Pitch Wheel (AbsPwl)

This control source always responds to movement of your MIDI controller's Pitch Wheel, but makes the Pitch Wheel unipolar. Whereas pulling the Pitch Wheel fully down usually generates a control signal value of -1, this control source generates a value of +1 when the Pitch Wheel is pulled fully down.

38 Global ASR (GASR2)

When the Globals parameter on the COMMON page is turned on, ASR2 becomes global, and is labeled GASR2. The functions of ASRs are explained on page 6-42 of the *Musician's Guide*. This control source does not appear in the Control Source list for parameters whose functions are local.

39 Global FUN2 (GFUN2)

When the Globals parameter on the COMMON page is turned on, FUN2 becomes global, and is labeled GFUN2. The functions of FUNs are explained in Chapter 17 of the *Musician's Guide*. This control source does not appear in the Control Source list for parameters whose functions are local.

40 Global LFO (GLFO2)

When the Globals parameter on the COMMON page is turned on, LFO2 becomes global, and is labeled GLFO2. The functions of LFOs are explained on page 6-40 of the *Musician's Guide*. This control source does not appear in the Control Source list for parameters whose functions are local.

41 Global LFO Phase (GLFO2ph)

When the Globals parameter on the COMMON page is turned on, LFO2 becomes global, and is labeled GLFO2. The functions of LFOs are explained on page 6-40 of the *Musician's Guide*. This control source does not appear in the Control Source list for parameters whose functions are local.

42 Global FUN 4 (GFUN4)

When the Globals parameter on the COMMON page is turned on, FUN 4 becomes global, and is labeled GFUN4. This control source does not appear in the Control Source list for parameters whose functions are local.

43 Volume Control (VolCtl)

This control source will always respond to MIDI Controller 07 messages. Assign this value to a parameter when you want MIDI volume messages to affect the parameter.

44 Pan Control (PanCtl)

This control source always responds to MIDI Controller 10 messages. Assign this value to a parameter when you want MIDI pan messages to affect the parameter.

45 Balance Control (BalCtl)

This control source will always respond to MIDI Controller 08 messages. Assign this value to a parameter when you want MIDI balance messages to affect the parameter.

46 Channel Count (ChanCnt)

This control source keeps track of the total number of active voice channels (how many notes are playing), and converts the number into a control signal between 0 and +1. The control signal's value is 1 when all 48 voice channels are active, and 0 when no voice channels are active.

You can use this control source in several ways. One example is to limit the volume of each note so that you have a more nearly constant volume regardless of how many notes you're playing (this is independent of the effect of attack velocity on volume). To set this up, you would go to the F4 AMP page in the Program Editor, and set the Src1 parameter to a value of **ChanCnt**. Then set the Depth parameter to a negative value. This will decrease the overall amplitude of each note as you play more simultaneous notes. This example works best with short-release sounds. It's great for an organ program, for example.

Channel count is also useful for controlling the modulation applied to a sound. For example, you may have a sound that you use both as a lead and for rhythm. Suppose you want a deep vibrato when you're soloing, but less vibrato when you're playing chords. Set up the vibrato by using **LFO1** as the value for the Src2 parameter on the PITCH page in the Program Editor. Set the MinDpt parameter to **72 cts**, and the MaxDpt parameter to **12 cts**. Then set the value of the DptCtl parameter to **ChanCnt**, and You'll get maximum vibrato depth when only one note is active. (Channel count outputs a control signal of 0 when no notes are playing, so with only one note playing, its value is near 0, which causes the DptCtl parameter to generate a value near its minimum: 72 cents in this case.)

If you want to increase the depth of the vibrato as you increase the number of active notes, set the value of the MaxDpt parameter higher than that of the MinDpt parameter.

Note: There are no control sources that correspond to the numeric entries 47—54.

55 Sync State (SyncSt)

This unipolar control source responds to MIDI clock messages received from an external MIDI device. Sync State switches on (+1) at each clock start, and switches off (0) with each clock stop.

56 A Clock

This is a unipolar square wave that responds to MIDI clock messages. It switches to +1 and back to 0 with every clock beat. This control source looks first for externally received MIDI clock messages, and if none is received, it responds to the K2600's internal clock, which is always running. The internal clock speed is set with the Tempo parameter in Song mode.

57 Negative A Clock (~A Clock)

This is the opposite of A clock, that is, it switches from 0 to +1 with every clock beat (the square wave is 180 degrees out of phase with that of A Clock).

58 B Clock

This is similar to A Clock, but it's bipolar—it switches from +1 to -1 with every clock beat.

59 Negative B Clock (~B Clock)

The opposite of B Clock, this bipolar control source switches from -1 to +1 with every clock beat (the square wave is 180 degrees out of phase with that of B Clock).

60, 61 Global Phase 1 and 2 (G Phase 1, G Phase 2)

These bipolar global control sources are both rising sawtooth waves that rise from -1 to +1 with each MIDI clock beat. Like A Clock and B clock, they look for an external clock signal, and if none is received, they respond to the K2600's internal clock.

62, 63 Global Random Variant 1 and 2 (GRandV 1, GRandV 2)

These are also bipolar and global, and generate random control signal values between -1 and +1 when assigned to a control source parameter. There is a subtle difference in the randomness of the signals they generate, therefore choosing between them is a matter of preference.

96 Note State (Note St)

At any moment, any given note is either on or off; this is its Note State. Note State can be used as a unipolar control source that responds to each note that's played. It switches to +1 when the note starts, and stays on as long as the note is held on (by the sustain pedal, for example), or by holding down the trigger for that note. It switches to 0 when the note is no longer sustained by any means. For example, if you play a note, then hold it with the sustain pedal, its Note State is still on (+1) even if you've released the key that triggered the note. As soon as you release the sustain pedal, the note's Note State switches to off (0), even if it has a long release and you can still hear the release section of the note.

97 Key State (Key St)

This is a unipolar control source that responds to the motion of your MIDI source's keys (or other note trigger). It switches to +1 when a key is pressed, and switches to 0 when the key is released. Its effect differs from Note State in that when the key that switched it on is released, it will switch off even if the note is sustained. If you're using a non-keyboard MIDI source, Key State switches to 0 when the equivalent of a key release is sent.

98 Key Number (KeyNum)

This is a unipolar control source that generates its signal value based on the MIDI key number of each note triggered. That is, it generates a value of 0 in response to MIDI key number 0, a value of 64 in response to MIDI key number 64, and so on. Note that some parameters, such as Enable Sense on the Program Editor Layer Page, will not accept this parameter. GKeyNum, controller number 129, would be acceptable however.

99 Bipolar Key Number (BKeyNum)

This is like KeyNum, but generates a signal value of -1 in response to MIDI key number 0, a value of 0 in response to MIDI key number 64, and a value of +1 in response to MIDI key number 127.

100 Attack Velocity (AttVel)

This unipolar control source responds to Attack velocity values received at the K2600's MIDI In port. Velocity values of 0 cause it to generate a signal value of 0, while velocity values of 127 will generate a value of +1. All other velocity values will result in signal values proportionally scaled between 0 and +1. Note that some parameters, such as Enable Sense on the Program Editor Layer Page, will not accept this control source. GAttVel, controller number 130, would be acceptable however.

101 Inverse Attack Velocity (InvAttVel)

This is the opposite of AttVel, generating a signal value of 0 in response to attack velocity values of 127.

102 Polyphonic Pressure (PPress)

This unipolar control source responds to poly pressure (aftertouch) messages received via MIDI. It generates a signal value scaled from 0 to +1 based on the poly pressure value range of 0—127.

103 Bipolar Polyphonic Pressure (BPPress)

This is like PPress, but scales its signal value from -1 to +1.

104 Release Velocity (RelVel)

Also unipolar, this control source scales its signal value from 0 to +1 in response to release velocity values from 0—127.

105 Bipolar Attack Velocity (Bi-AVel)

This is similar to AttVel, but scales its signal values from -1 to +1.

106, 107 Velocity Triggers 1 and 2 (VTRIG1, VTRIG2)

These unipolar control sources are switch controls, that is, they generate signal values of either 0 or +1. These must be programmed in order to have an effect; their programming parameters are found on the VTRIG page in the Program Editor. When a VTRIG's Sense parameter is set to normal, it switches to +1 when a note plays at a dynamic level exceeding the dynamic level set for its Level parameter. See page 6-44 of the *Musician's Guide* for more information.

108, 109 Random Variants 1 and 2 (RandV1, RandV2)

These are similar to GRandV1 and GRandV2, but are local, so will affect each control source parameter independently.

110, 111 ASR1, ASR2

These are programmable envelopes with three segments, Attack, Sustain, and Release. Their control source signals are unipolar. See page 6-42 of the *Musician's Guide* for a thorough explanation.

112, 113 FUN1, FUN2

These generate their control source signals by combining the control signal values of two programmable inputs, and performing a mathematical function on the result. Their control signals can be unipolar or bipolar, depending on the control sources assigned as their inputs. See page 6-43 of the *Musician's Guide*. FUN2 becomes global (GFUN2) when the Globals parameter on the COMMON page in the Program Editor is set to **On**.

114 LFO1

LFO1 can be unipolar or bipolar depending on the value set for the Shape parameter on its programming page. See page 6-40 of the *Musician's Guide*.

115 LFO1 Phase (LFO1ph)

This bipolar control source generates it signal based on the cycle of LFO1. When the phase of LFO1 is 0 degrees, the signal value of LFO1ph is 0. When the phase of LFO1 is 90 degrees, the signal value of LFO1ph is 1. When the phase of LFO1 is 180 degrees, the signal value of LFO1ph is 0. When the phase of LFO1 is 270 degrees, the signal value of LFO1ph is -1.

116 LFO2

This functions exactly the same as LFO1, when the Globals parameter is set to **Off** (on the COMMON page in the Program Editor). When the Globals parameter is set to **On**, LFO2 becomes global (GLFO2).

117 LFO2 Phase (LFO2ph)

This functions exactly the same as LFO1ph, responding to the cycle of LFO2.

118, 119 FUN3, FUN4

These function exactly the same as FUNs 1 and 2, when the Globals parameter is set to **Off** (on the COMMON page in the Program Editor). When the Globals parameter is set to **On**, FUN4 becomes global (GFUN4).

120 Amplitude Envelope (AMPENV)

This programmable unipolar control source lets you vary the effect of a control source parameter over time. See page 6-35 of the *Musician's Guide*.

121, 122 Envelopes 2 and 3 (ENV2, ENV3)

These are programmed in the same way as AMPENV, but they can be bipolar.

123 Loop State (Loop St)

This unipolar control source switches to +1 when the currently playing sample reaches its LoopStart point. If you've programmed a sound with a User amplitude envelope, Loop St will always be on (+1) for that sound. See page 14-17 of the *Musician's Guide* for more about sample loops.

124 Sample Playback Rate (PB Rate)

The signal value of this bipolar control source is determined by the sample playback rate of each note. The playback rate is a function of the amount of transposition applied to a sample root to play it at the proper pitch for each note. If you trigger a note where a sample root is assigned, the PB Rate signal value for that note is 0. If the note is above the sample root, the sample is transposed upward, and its playback rate is higher than that of the sample root. Consequently the PB Rate signal value for that note will be positive. If the note is below the sample root, the PB Rate signal value will be negative.

125 Attack State (Atk State)

This unipolar control source switches to +1 and back to 0 very quickly with each note start.

126 Release State (Rel State)

This unipolar control source switches to +1 when a note is released, and stays on until the note has completed its release (faded to silence), then it switches to 0. It will stay on if a note is sustained, even if its trigger (key, string, whatever) is released.

127 ON

This generates a constant control signal value of +1.

128 -ON

This generates a constant control signal value of -1 (the numeric entry 128 selects a value of **OFF** in the MIDI Control Source list).

129 GKeyNum

Uses the key number (global) to modify whatever it is patched into. Higher notes will have a very different effect than will lower notes. Users can use this new Source to control any K2600 parameters, or to scale amplitude or pitch.

130 GAttVel

This is updated every time you strike another key (kind of a multi- trigger function).

In addition to enabling (triggering) layers from any controller (works like an on/off switch), you can set the assigned controller’s threshold (value, or range of values from 0-127), thus defining the controller’s active range where it will enable the layer.

For example, you could create a 32-layer nylon guitar in which each layer is assigned to a different VAST algorithm and each layer is enabled by discrete narrow velocity ranges. This would produce 32 different sounding layers with 32 cross switch points emulating a picked guitar where no two attacks are exactly alike. If the layers' velocity ranges were very close together yet not overlapping, you could create very subtle nonrepeating changes. This kind of power usually eludes most sample playback devices, as this technique uses only one layer of polyphony, due to cross switching versus cross fading.

131, 132 GHiKey, GLoKey

These control sources work the same as GKeyNum except that they track the highest key currently held and the lowest key currently held respectively. By using one of these as the only source for pitch tracking, you can create monophonic-like layers within a polyphonic program.

Constant Control Sources

The remaining control sources are constants, which appear only when you're assigning control sources as inputs for the FUNs. Assigning one of these values fixes the input's control signal value at a steady level.

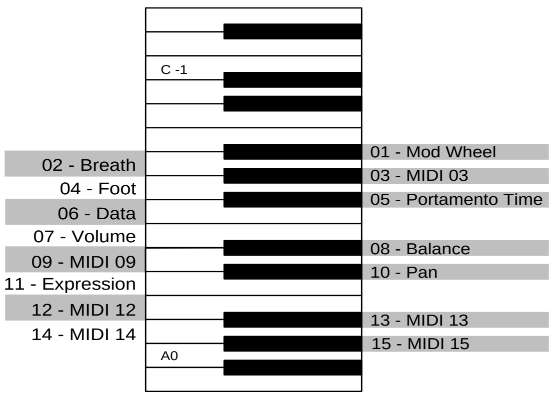
Assigned Value	Corresponding Constant	Assigned Value	Corresponding Constant
133	-0.99	201	0.09
134	-0.98	202	0.10
135	-0.97	203	0.12
136-140	-0.96 to -0.92	204	0.14
141	-0.91	205	0.16
142	-0.90	206-210	0.18 to 0.26
143-145	-0.88 to -0.84	211-215	0.28 to 0.36
146-150	-0.82 to -0.74	216-220	0.38 to 0.46
151-155	-0.72 to -0.64	221-225	0.48 to 0.56
156-160	-0.62 to -0.54	226-230	0.58 to 0.66
161-165	-0.52 to -0.44	231-235	0.68 to 0.76
166-170	-0.42 to -0.34	236-240	0.78 to 0.86
171-175	-0.32 to -0.24	241	0.88
176-180	-0.22 to -0.14	242	0.90
181	-0.12	243	0.91
182	-0.10	244	0.92
183	-0.09	245	0.93
184	-0.08	246-250	0.94 to 0.98
185	-0.07	251	0.99
186-190	-0.06 to -0.02	256	OFF
191	-0.01		
192	0.00		
193	0.01		
194	0.02		
195	0.03		
196-200	0.04 to 0.08		

Note: There are no control sources that correspond to numeric entries 252—254.

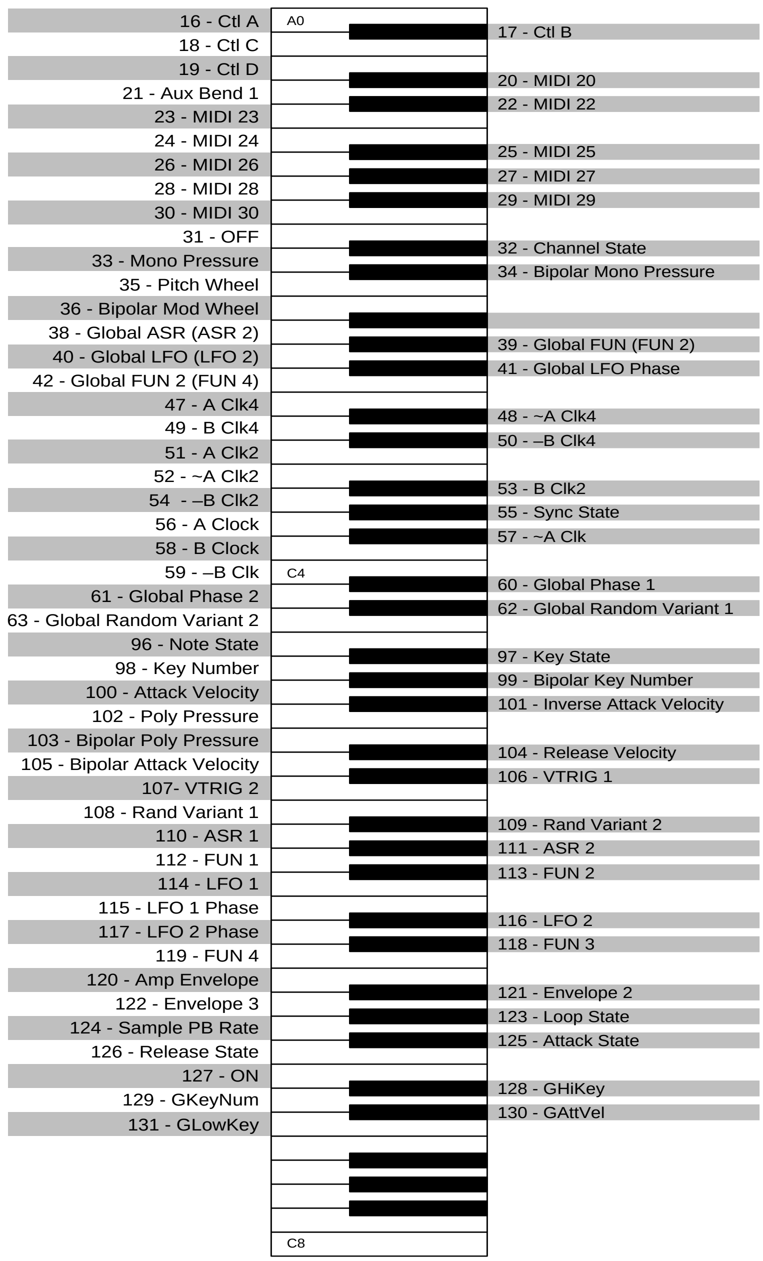
Keyboard Shortcuts for Control Sources

You can use the keyboard of your MIDI source to choose control sources, since most key numbers correspond to a value on the Control Source list. If you have a certain control source that you use over and over (for example, LFO1), this can be the quickest way to enter its value. To do this, highlight a parameter that uses a value from the Control Source list, hold down **Enter**, then strike the key corresponding to the control source you want to choose. LFO1, for example, corresponds to B5.

C-1 to A0
(Below Standard 88-note Keyboard)



A0 to C8
(Standard 88-note Keyboard)



Chapter 5
MIDI Note Numbers

K2600 Note Numbers and MIDI Note Numbers

K2600	MIDI
C -1–B -1	0–11
C 0–B 0	12–23
C 1–B 1	24–35
C 2–B 2	36–47
C 3–B 3	48–59
C 4 (Middle C)–B 4	60–71
C 5–B 5	72–83
C 6–B 6	84–95
C 7–B 7	96–107
C 8–B 8	108–119
C 9–G 9	120–127

You can assign samples to keymaps in the range from C 0 to G 9. The K2600 will respond to MIDI events in the octave from C -1 to B -1. If a Note On event is generated in the range from C -1 to B -1, the K2600 will respond by setting the Intonation key correspondingly (C -1 will set it to C, C$^{\#}$ -1 will set it to C$^{\#}$, etc.)

Note Numbers for Percussion Keymaps

Most of the K2600’s percussion programs have keymaps that place the various percussion timbres at standardized key locations. There are eight drum keymaps: Preview Drums, five 5-octave kits (two dry and three ambient), a 2-octave kit, and the General MIDI kit. The keymap **30 General MIDI Kit** adheres as closely as possible to the General MIDI standard for placement of timbres. As a rule, programs that use this keymap can be assigned in percussion tracks for prerecorded sequences and will play appropriate timbres for all percussion notes.

The timbres are located consistently within the 5-octave kit keymaps so you can interchange keymaps within percussion programs freely without changing the basic timbres assigned to various notes (snare sounds will always be at and around Middle C, for example). The note assignments for the timbres in the 5-octave kit and 2-octave kit keymaps are listed below. MIDI note number 60 (Middle C) is defined as C 4.

5-Octave Percussion Keymaps (Range: C2–C7)

MIDI Note Number	Key Number	Sample Root
36-37	C2-C#2	Low Tom
38-39	D2-D#2	Low Mid Tom
40-41	E2-F2	Mid Tom
42-43	F#2-G2	Hi MidTom
44-45	G#2-A2	Mid Hi Tom
46	A#2	Hi Tom
47–51	B 2–D# 3	Kick
52–54	E3–F#3	Snare (Sidestick)
55-56	G3-G#3	Low Snare (dual vel. on Dry Kit 1)
57-59	A3-B3	Mid Snare (dual vel. on Dry Kit 1)
60-61	C4-C#4	Hi Snare (dual vel. on Dry Kit 1)
62–64	D 4–E 4	Closed HiHat
65–67	F 4–G 4	Slightly Open HiHat
68–69	G# 4–A 4	Open HiHat
70–71	A# 4–B 4	Open to Closed HiHat
72	C 5	Foot-closed HiHat
73-74	C#5-D5	Low Crash Cymbal
75-78	D#5-F#5	Pitched Crash Cymbals
79	G5	Splash Cymbal
80	G#5	Ride Cymbal (Rim)
81-82	A5-A#5	Ride Cymbal (Rim and Bell)
83-84	B5-C6	Ride Cymbal (Bell)
85	C# 6	Cowbell
86	D 6	Handclap
87	D# 6	Timbale
88	E 6	Timbale Shell
89	F 6	Conga Tone
90	F#6	Conga Bass Hi
91	G 6	Conga Slap
92	G#6	Conga Bass Low
93	A 6	Clave
94	A# 6	Cabasa
95–96	B 6–C 7	Tambourine Shake

2-Octave Percussion Keymaps (Range: C3 - C5)

MIDI Note Number	Key Number	Sample Root
48–49	C 3–C$^{\#}$ 3	Kick
50	D 3	Low Tom
51	D$^{\#}$ 3	Cowbell
52	E 3	Low Tom
53	F 3	Mid Tom
54	F$^{\#}$ 3	Cowbell
55	G 3	Mid Tom
56	G$^{\#}$ 3	Timbale
57	A 3	High Tom
58	A$^{\#}$ 3	Snare (Sidestick)
59	B 3	High Tom
60-61	C4-C$^{\#}$4	Snare (dual velocity)
62	D 4	Closed HiHat
63	D$^{\#}$4	Ride Cymbal (Rim and Bell)
64	E 4	Closed HiHat
65	F 4	Slightly Open HiHat
66	F$^{\#}$ 4	Crash Cymbal
67	G 4	Slightly Open HiHat
68	G$^{\#}$ 4	Crash Cymbal
69	A 4	Open HiHat
70	A$^{\#}$ 4	Crash Cymbal
71	B 4	Open to Closed HiHat
72	C 5	Foot-closed HiHat

Chapter 6
MIDI, SCSI, and Sample Dumps

SCSI Guidelines

The following sections contain information on using SCSI with the K2600, as well as specific sections dealing with the Mac and the K2600.

Disk Size Restrictions

The K2600 accepts hard disks with up to 2 gigabytes of storage capacity. If you attach an unformatted disk that is larger than 2 gigabytes, the K2600 will still be able to format it, but only as a 2 gigabyte disk. If you attach a *formatted* disk larger than 2 gigabytes, the K2600 will not be able to work with it; you could reformat the disk, but this—of course—would erase the disk entirely.

Configuring a SCSI Chain

Here are some basic guidelines to follow when configuring a SCSI chain:

1. According to the SCSI Specification, the maximum SCSI cable length is 6 meters (19.69 feet). You should limit the total length of all SCSI cables connecting external SCSI devices with Kurzweil products to 17 feet (5.2 meters). To calculate the total SCSI cable length, add the lengths of all SCSI cables, plus eight inches for every external SCSI device connected. No single cable length in the chain should exceed eight feet.

2. The first and last devices in the chain must be terminated. There is a single exception to this rule, however. A K2600 with an internal hard drive and no external SCSI devices attached should have its termination disabled. If you later add an external device to the K2600’s SCSI chain, you must enable the K2600’s termination at that time.

   There’s a switch on the rear panel of the K2600, which you can use to disable the K2600’s termination. We recommend, however, that you leave this switch set to **Auto**, which enables the K2600 to switch termination on or off depending on your SCSI configuration.

   Poor termination is a common cause of SCSI problems. Having more than two terminators on the bus will overload the bus drivers, but this should not cause permanent damage to the hardware. Poor termination can corrupt the data on your disk, however, as can bad SCSI cables.

   A note about active termination: The K2600 uses active termination of the SCSI bus. Active termination has some benefits over traditional passive termination. Some people view active termination as a cure for all SCSI problems, but this isn’t true. Active terminators are appropriate at the end of a SCSI chain. All APS SR2000-series external drives use internal active termination that can be switched on or off.

3. Each device in the chain (including internal hard drives) must have its own unique SCSI ID. The default K2600 ID is 6. Macintoshes® use 7 and 0.

4. Use only true SCSI cables: high quality, twisted pair, shielded SCSI cable. Do not use RS432 or other nonSCSI cables.

The majority of SCSI cables we've tested were poorly made and could damage data transferred to and from the disk. Nearly all the SCSI data problems Young Chang's engineering department has encountered have been due to bad cables that didn't twist pairs of wires properly. Correctly made SCSI cables have one ground wire for every signal wire and twist them together in signal/ground pairs. Cables made by APS Technologies (800-233-7550) are very good and are highly recommended. Young Chang manufactures 1 and 2 meter 25-25 SCSI cables, that we can also recommend. Good cables are essential to reliable data transfers to and from the disk drive.

5. You should buy all SCSI cables from a single source to avoid impedance mismatch between cables.
6. Theoretically all eight SCSI IDs can be used. However, feedback from users has shown us that many people have problems with more than five or six devices in a chain. If you have seven or eight devices and are having problems, your best bet is to make sure you have followed all of the previous information, especially with respect to cables.
7. Connect all SCSI cables before turning on the power on any equipment connected by SCSI cables. Plugging or unplugging SCSI cables while devices are powered on can cause damage to your devices or instrument.
8. When using a Macintosh, power up the K2600 and other devices first.
9. The K2600 file format is a proprietary format; no other device will be able to read or write a Kurzweil file.
10. The floppy disk format of the K2600 is DOS. The SCSI disk format is a proprietary form that is close to DOS, but it is not DOS. Nonetheless, the K2600 can read from and write to the first partition on a DOS-formatted disk.
11. You can view, copy, move, name, and delete files on a K2600-formatted floppy disk or removable media hard drive, with a PC or Macintosh running a DOS mounting utility program such as Access PC.
12. As long as the SCSI bus is properly terminated there is no way you can damage your hardware simply by operating it. There are a few hazards you should be aware of, however:

    The only damage that usually occurs to SCSI hardware comes from static electricity discharging to SCSI connector pins when the cables are disconnected. The silver colored shell of the SCSI connector on the end of the cable is connected to ground and is safe to touch, but the brass colored pins inside eventually lead to the SCSI interface chip and are vulnerable. You should discharge static from your body before touching SCSI connectors, by touching the 1/4-inch jacks on the rear of the K2600 or another grounded metal object. Any devices connected to the SCSI bus should be turned off when plugging or unplugging SCSI cables.

    If the K2600 is connected to a Macintosh or PC you should make sure that the computer cannot access a SCSI disk at the same time the K2600 does (see below for more information on this). If you occasionally want to share a drive, but don't want to take any risks, you should connect and disconnect devices as needed. If you want to share drives often and cannot constantly disconnect and reconnect devices, make sure the Mac or PC is really done with the disk before using the K2600. Furthermore, you should quit or exit from all running programs and disable screen savers, email, network file sharing, and any INITs or TSRs that run in the background. If the computer and K2600 access the disk at the same time there will be no damage to the hardware, but the bits on the disk, K2600, and computer memory can easily be corrupted. You may not know that damage has been done to these bits until unexpected things start to happen for no apparent reason.

13. A good way to verify your SCSI hookup is to save and load some noncritical files.

K2600 and Macintosh Computers

There are several points to consider when using a Macintosh with the K2600:

1. The Mac is not well equipped for having another SCSI master on the bus (that is, the K2600). It assumes that it owns the bus and its drives—consequently it will not allow the K2600 to address any of its drives. Therefore, you should not attempt to read from or write to any drive mounted on the Mac's desktop. Even more fundamental is the problem that the Mac assumes that the bus is always free, so if it tries to do anything via SCSI when the K2600 is doing anything via SCSI, you'll have problems. The only solution is to wait until your Mac is completely idle before accessing SCSI from the K2600.

2. The Mac and the K2600 cannot share a drive in any way, with or without partitions. If you are using a removable-media drive (like a Syquest or Zip drive), you can't easily use it for both Mac-formatted disks and K2600-formatted disks. To prevent problems, you will need to unmount the drive from the Mac desktop before using a K2600-formatted disk in the drive. The Mac will basically ignore the disk if it's not in Mac format, but once you insert a Mac-formatted volume, the Mac owns it. Don't forget: inserting a disk in a removable drive will cause the Mac to access SCSI, so don't try to use the K2K at that moment.

3. The only good reason for connecting the Mac and the K2600 on the same SCSI bus is to use Alchemy or the equivalent. If you're using a patch editor or librarian, you can connect via MIDI. Connecting via SCSI will allow fast sample transfers through the SMDI protocol. In this type of configuration the easiest solution is to let the K2600 have its own drive, and the Mac have its own drive.

   However, we have discovered that when using a K2600 with a Mac and a removable media drive in the middle of the chain, the following scenario will work:

   Start with a Mac-formatted disk in the drive. When you want to use the K2600, put the drive to sleep from the K2600. You can then change to a K2600-formatted disk and perform whatever disk operations you need. When you want to go back to the Mac, put the drive to sleep again, switch disks, and then wake up the drive by pressing **Load**. Of course the K2600 will tell you it can't read the disk, but the Mac will be able to.

Accessing a K2600 Internal Drive from the Mac

Access PC is one of the many programs for the Mac that allow it to format, read, and write to DOS floppy disks and removable SCSI cartridges. Reading and writing to an internal hard disk on the K2600 is fine, but don't try to format it using Access PC on a Mac.

If you use a Mac with Access PC to address your K2600's internal hard disk, never save or delete files from the K2600 when the internal disk is mounted by the Mac. This could result in corrupted files or directories—it could even corrupt the entire disk. Access PC has no way of knowing when the K2600 has modified the disk contents, and it could write over existing data, or crash while trying to read data that are no longer there. The safest approach is to connect a drive to either the K2600 or the Mac, but not to both at the same time. Of course, you can't always predict when a Mac will access its drive, and it doesn't do SCSI bus arbitration, so using the Mac while using the SCSI bus from the K2600 (for example, doing a Disk-mode operation) is also a bad idea, and can cause the Mac to hang.

The MIDI Sample Dump Standard

Samples can be transferred between the K2600 and most other samplers and computer sampling programs using the MIDI Sample Dump Standard.

Due to the relatively slow transfer rate of MIDI data, transferring samples into the K2600 via the MIDI Sample Dump Standard can take a long time, on the order of a coffee break for a large sample. Most samplers, synthesizers, and software will “freeze up” during this process, preventing other features of the machine or program from being used. Your K2600, however, will allow you to continue playing the instrument or using any of its sound editing features during a MIDI Sample Dump! The transfer takes place in the background; the MIDI-mode LED on the K2600’s front-panel flashes repeatedly during the transfer, so you will always know if the MIDI Sample Dump is proceeding. The MIDI-mode LED flashes only when the K2600 is transmitting or receiving a MIDI Sample Dump, or when it receives a MIDI System Exclusive message.

Note: if you’re using Sound Designer® to transfer samples, you’ll have to offset the sample number by 2 to transfer the right sample. For example, if you want to dump sample ID 208 from the K2600, then when you begin the sample fetching command from Sound Designer, instruct it to get sample 210.

Loading Samples with the MIDI Standard Sample Dump

To load a sample into the K2600 from an external source such as a computer or sampler, first connect the MIDI Out port of the sampler (or computer) to the K2600’s MIDI In port, and connect the K2600’s MIDI Out to the MIDI In of the sampler. This is known as a MIDI loop.

Next, access the Sample Dump facility on the sampler. In addition to selecting which sample you wish to transfer over MIDI, you will need to set the correct sample dump channel number and destination sample number. The channel number should match the K2600’s SysEx ID parameter (on the RECEIVE page in MIDI mode). If the sampler has no facility for setting the Sample Dump channel number, try setting the K2600’s SysEx ID parameter to **0** or **1**. Alternatively, if you set the SysEx ID to **127**, the K2600 will accept a MIDI Sample Dump no matter what Sample Dump channel is used to send the sample dump.

If the sampler has a provision for setting the destination sample number, you can use it to specify the ID the K2600 will use for storing the sample. The K2600 sample number is mapped from the destination sample number as follows:

Sample Number	K2600 ID
0	uses lowest unassigned ID between 200 and 999.
1-199	adds 200 to the ID (for example, 5 becomes 205 in the K2600.)
200-999	ID is the same number.

If the sample number maps to a number already assigned to a RAM sample in the K2600, the RAM sample will be deleted before the K2600 loads the new sample. The K2600 will always map sample number zero to an unassigned ID, and therefore no samples will be overwritten when zero is specified.

Some computer-based sample editing software limits the sample numbers to a low range such as 1-128. This conflicts with the K2600, which reserves IDs 1-199 for ROM samples, which cannot be loaded or dumped. To get around this, the K2600 adds 200 to any numbers between 1 and 199. Therefore, if you want to load a sample into the K2600 at number 219, but your program can't transfer samples at numbers greater than 128, specify number 19 (There's an exception to this; please see *Troubleshooting a MIDI Sample Dump* on page 6-6).

At this point, you're ready to try loading a sample. See *Accessing a New K2600 Sample* on page 6-6 to learn how to use samples once they've been dumped to the K2600.

Getting a Sample into a Sample Editor from the K2600

Connect the MIDI ports of the K2600 and the computer/sampler in a MIDI loop as described for the Sampler/Computer to K2600 procedure above.

Access the computer software's "Get Sample" page (it might be called something different). As with loading a sample into the K2600, the K2600 adds 200 to dump request sample numbers between 1 and 199. K2600 samples with IDs from 1 to 199 are ROM samples, and cannot be dumped. Therefore, if you want to get sample number 219 from the K2600 but your program can't transfer samples at numbers greater than 128, specify number 19 (There's an exception to this; please see *Troubleshooting a MIDI Sample Dump* on page 6-6).

Loading a Sample into the K2600 from another K2600

Connect the MIDI ports of the two K2600s in a MIDI loop as described for the Sampler/Computer to K2600 procedure above.

On the source K2600, go to the Sample Editor and select the sample you wish to transfer. To do this, start in Program mode and press **Edit**, followed by the **KEYMAP** soft button. Now you should be on the KEYMAP page. Now move the cursor to the Sample parameter, use any data entry method to select the desired sample, then press **Edit**.

To start the sample transfer, press the **Dump** soft button. A dialog will appear, suggesting the ID for the sample to be dumped to the destination K2600. The source K2600 will suggest the same ID as it uses for the sample, but you can change the destination ID with any data entry method. If you choose the default by pressing **Yes**, the sample will transfer to the same ID on the destination K2600 as it is on the source K2600.

Dumping from the K2600 to a Sampler

This procedure is the same as dumping a sample from one K2600 to another. This will work only if the sampler supports the MIDI Sample Dump Standard.

Dumping a Sample from the K2600 to a MIDI Data Recorder

This can be accomplished by connecting the MIDI Out port of the K2600 to the MIDI In port of the MIDI Data Recorder. Go to the Sample Editor and select the K2600 sample you wish to transfer. Set up the MIDI Data Recorder to begin recording, and press the **Dump** soft button on the Sample Editor page. This will bring up a dialog allowing you to change the sample number in the dump if you wish. In most cases, you will just use the default value. The K2600's MIDI mode LED will flash while the data transfer is in progress.

Loading a Sample into the K2600 from a MIDI Data Recorder

Connect the MIDI Out port of the Data Recorder to the MIDI In port of the K2600. Load the appropriate file containing the MIDI Sample Dump data into the Data Recorder, and send the file. The K2600's MIDI mode LED will flash during this procedure.

Accessing a New K2600 Sample

First, select the K2600 program you wish to play the new sample from, and press **Edit**. Then select the layer you wish (using the **Chan/Bank** buttons if necessary), press the **KEYMAP** soft button, and select a keymap. Use the default keymap called **168 Silence** if you don't want to alter any existing keymaps.

Now, enter the Keymap Editor by pressing **Edit** once again. Use the Sample parameter to select the new sample. If the new sample was loaded from another K2600, it will have the same ID as it did on the other K2600. If the sample was loaded from any other source, its ID will be defined as described in *Loading Samples with the MIDI Standard Sample Dump* on page 6-4).

The name of the sample will be assigned by the K2600 if the sample has been assigned to a previously unused ID. In most cases, the sample will have a name of **New Sample - C 4**.

The name will be **New Sample! - C 4** (note the exclamation point) if checksum errors were detected by the K2600. Checksum errors are usually not serious, since they may just mean the source sampler doesn't adhere to the MIDI Sample Dump Standard checksum calculation. In other cases, a checksum error could indicate that the MIDI data flow was interrupted during the sample transfer.

You can now press **Edit** to edit the parameters of the new sample such as Root Key, Volume Adjust, Pitch Adjust, and Loop Start point. You can also rename the sample. Be sure to save the parameters you change when you press **Exit**. Once the sample is adjusted to your liking, you can assign it to any Keymap.

Troubleshooting a MIDI Sample Dump

This section will help you identify what has gone wrong if your MIDI sample dumps fail to work.

When Loading Samples to the K2600

There are two reasons a K2600 will not accept a MIDI Sample Dump. First, a dump will not be accepted if the destination sample number maps to a K2600 sample that is currently being edited—that is, if you're in the Sample Editor, and the currently selected sample has the same ID as the sample you're trying to dump. Second, a dump will not be accepted if the length of the sample to be dumped exceeds the available sample RAM in the K2600. There may be samples in the K2600 RAM that you can save to disk (if not already saved) and then delete from RAM to free up sample RAM space. You can delete the current sample by pressing the Delete soft button while in the Sample Editor.

Note that when you're loading a sample to an ID that's already in use, the K2600 will not accept a MIDI Sample Dump if the length of the sample to be loaded exceeds the amount of available sample RAM *plus* the length of the existing sample. If the K2600 accepts the sample load, the previously existing sample will be deleted.

Also note that certain computer-based editing programs will subtract one from the sample number when performing MIDI sample transfers to remote devices. So if you instruct these programs to send a sample to the K2600 as sample ID 204, the program will send the sample as 203. The only way to know if your program behaves in this manner is to try a MIDI Sample Dump and see what happens.

When Dumping Samples From the K2600

Certain computer-based sample editing programs subtract one from the sample number when performing MIDI Sample transfers to remote devices. For instance, if you tell these programs to get sample number 204, the programs will request that the K2600 dump sample ID 203, which would ordinarily dump a different sample from the one you intended, possibly causing the dump to fail. The K2600 automatically counteracts this offset by adding a number to sample requests. This was done because more sample editing programs create this offset than do not. If you find that the K2600 is sending samples with higher IDs than the ones you requested, you can compensate by requesting the sample ID one lower than the one you want. For example, if you want the K2600 to dump sample 205, ask for sample 204.

Some samples in the K2600 are copy-protected. These include all ROM samples and possibly some third-party samples. The K2600 will not dump these samples.

Aborting a MIDI Sample Dump

The **Abort** soft button in the Sample Editor can be used to cancel any sample load into the K2600 from an external source (for example, a computer or a sampler). This button will also halt a sample dump from the K2600. The K2600 will ask for confirmation before it aborts the sample dump.

SMDI Sample Transfers

You can use Passport's Alchemy® and Opcode's Max® SMDI-capable Macintosh® software packages to transfer mono and stereo samples to and from the K2600. These applications use the SMDI data transfer format (SMDI stands for SCSI Musical Data Interchange—pronounced *smiddy*. SMDI is parallel, not serial, so sample transfers can be made much faster than with the MIDI sample dump standard.

Each of these applications has commands for getting and sending samples, which is how you'll make the transfer from your offline storage to the K2600. Once the samples have been loaded to the K2600, you can use the Keymap and Sample Editors as you would with any other sample. Check your manuals for Alchemy or Max for the specifics.

Keep in mind that when transferring samples via SMDI, the K2600's sound engine is disabled, so you can't play it during a SMDI transfer as you can during a MIDI sample transfer. The average SMDI sample transfer time is about 20K per second.

Chapter 7
System Exclusive Protocol

K2600 System Exclusive Implementation

The MIDI System Exclusive capabilities of the K2600 allow you to manipulate objects in the K2600's memory from a computer system, another K2600, or a MIDI data recorder. The following is a reference to the SysEx protocol used by the K2600. This information can be used to build a simple object librarian software program. A word of advice—before you begin experimenting with SysEx, make sure you have saved anything of value in RAM to disk.

***NOTE**: To support new features and changes in the K2600 line of products, the internal program structure has been changed from that of the K2000. Due to these changes, you cannot transfer a K2000 program to a K2600, or a K2600 program to a K2000 via MIDI system exclusive. The K2600 software will continue to be enhanced, and in the future the K2600 will be capable of accepting K2000 programs over MIDI. As a result of this, computer based K2000 editor/librarians will not currently work with the K2600, unless they have been revised to accommodate the changes.*

Common Format

In the following discussion, the fields of the K2600 System Exclusive Protocol messages are notated as field(length), where field is the name of the particular information field in the message, and (length) is either 1, 2, 3, or n, representing the number of sequential MIDI bytes that make up the field. A length of **n** means that the field is of a variable length that is determined by its contents or subfields.

All K2600 SysEx messages have the common format:

sox(1) kid(1) dev-id(1) pid(1) msg-type(1) message(n) eox(1)

sox is always F0h, and represents start of System Exclusive.

kid must be 07h, and is the Kurzweil Manufacturer ID.

dev-id is Device ID. The K2600 will recognize a SysEx message if dev-id is the same is the SysEx ID parameter from the MIDI-mode RECEIVE page. If the K2600's SysEx ID parameter is set to 127, it will recognize SysEx messages no matter what the value of dev-id is.

pid is the Product Identifier, and must be 78h (120 decimal), indicating the SysEx message is for the K2600.

msg-type is the identifier of one of the K2600 SysEx messages defined below, and message is the variable-length message contents.

eox is always F7h, for end of System Exclusive.

Data Formats

K2600 SysEx messages are subdivided into fields that contain data in different formats. The various fields are shown in the Messages section below. Within a message, any fields for values that can be bigger than 7 bits are broken into 7 bit chunks. Thus two MIDI bytes gives 14 bits, three bytes gives 21 bits. The significant bits are right justified in the field. All bytes in a field must be present no matter what the value is. For example, an object type of 132 would be split into two MIDI bytes in a type field as 01 04:

decimal:	132
binary:	10000100
binary encoding for type(2) field:	0000001 0000100
decimal encoding for type(2) field:	1 4

Object name fields are sent as a string of ASCII values in a name field, with one MIDI byte of zero as a string terminator. For example, the name **Glass Kazoo** would be sent as follows:

	G	l	a	s	s	_	K	a	z	o	o	<null>
hex encoding for name field:	47	6C	61	73	73	20	4B	61	7A	6F	6F	00

Data sizes and offsets are sent in the size and offs fields.These values refer to quantities of 8-bit bytes in the K2600's memory, which is packed in the data field.

Binary data in the data field are sent in one of two formats, according to the value of the form field. If the form field equals zero, the data are transmitted as 4 bits or one "nibble" in every MIDI byte. If the form field equals one, then the data are sent as a compressed bit-stream, with 7 bits per MIDI byte. The bit-stream format is more efficient for data transmission, while the nibble format is easier to read (and write software for).

For example, to send the following four K2600 data bytes,

hex:	4F	D8	01	29
decimal:	79	216	1	41
binary:	01001111	11011000	00000001	00101001

eight MIDI bytes are sent in "nibble" format:

hex	04	0F	0D	08	00	01	02	09
decimal	4	15	13	8	0	1	2	9
binary	0000100	0001111	0001101	0001000	0000000	0000001	0000010	0001001

five MIDI bytes are sent in bit-stream format:

hex:	27	76	0	12	48
decimal:	39	118	0	18	72
binary:	0100111	1110110	0000000	0010010	1001000

The bit-stream format can be thought of as taking the binary bits of the K2600 data and, starting from the left, slicing off groups of 7 bits. Note that the trailing bits are set to zero.

After the data field, there is another field, xsum. This is a checksum field that is calculated as the least significant 7-bits of the sum of all of the MIDI bytes that make up the data field.

Messages

This section defines the K2600 System Exclusive message formats. Each message has a message type, which goes in the msg-type field (see *Common Format* on page 7-1), followed by the field definitions of the message.

DUMP = 00h type(2) idno(2) offs(3) size(3) form(1)

Requests the K2600 to send a data dump of an object or portion thereof. type and idno identify the object. offs is the offset from the beginning of the object's data; size describes how many bytes should be dumped starting from the offset. form indicates how the binary data are to transmitted (0=nibblized, 1=bit stream). The response is a LOAD message:

LOAD = 01h type(2) idno(2) offs(3) size(3) form(1) data(n) xsum(1)

This writes data into the specified object, which must exist. Both load and dump operate on the object data only. The response to a load message will be the following:

DACK = 02h type(2) idno(2) offs(3) size(3)

Load accepted, or

DNAK = 03h type(2) idno(2) offs(3) size(3) code(1)

Load not accepted. The code field indicates the cause of the failure, as follows:

Code	Meaning
1	Object is currently being edited
2	Incorrect checksum
3	ID out of range (invalid)
4	Object not found (no object with that ID exists)
5	RAM is full

To request information about an object, use:

DIR = 04h type(2) idno(2)

The type and idno identify the object. The response is an INFO message:

INFO = 05h type(2) idno(2) size(3) ramf(1) name(n)

This is the response to DIR, NEW, or DEL. If object is not found, size will be zero and name will be null. ramf is 1 if the object is in RAM.

NEW = 06h type(2) idno(2) size(3) mode(1) name(n)

Creates a new object and responds with an INFO message of the created object. The object's data will not be initialized to any default values. If idno is zero, the first available object ID number will be assigned. If mode is 0, the request will fail if the object exists. If mode is 1, and the object exists in ROM, a RAM copy will be made. If mode is 1, and the object exists in RAM, no action is taken.

DEL = 07h type(2) idno(2)

Deletes an existing object and responds with an INFO message for the deleted object. If there is only a RAM copy of the object, the response will indicate that the object doesn't exist anymore. However, if the deletion of a RAM object uncovers a ROM object, the INFO response will refer to the ROM object. A ROM object cannot be deleted.

CHANGE = 08h type(2) idno(2) newid(2) name(n)

Changes the name and/or ID number of an existing object. If newid is zero or newid equals idno, the ID number is not changed. If newid is a legal object id number for the object's type, then the existing object will be relocated in the database at the new ID number. This will cause the deletion of any object which was previously assigned to the newid. If the name field is null, the name will not change. Otherwise, the name is changed to the (null-terminated) string in the name field.

WRITE = 09h type(2) idno(2) size(3) mode(1) name(n) form(1) data(n) xsum(1)

Writes an entire object's data directly into the database. It functions like the message sequence DEL followed by NEW followed by a LOAD of one complete object data structure. It first deletes any object already existing at the same type/ID. If no RAM object currently exists there, a new one will be allocated and the data will be written into it. The object name will be set if the name string is non-null. The response to this message will either be a DACK or a DNAK, as with the load message. The offs field of the response will be zero. The K2600 will send a WRITE message whenever an object is dumped from the front-panel (using a **Dump** soft button), or in response to a READ message.

The mode field is used to determine how the idno field is interpreted.

If mode = 0, the idno specifies the absolute ID number to write to, which must exist (must be valid). If idno equals zero, write to the first available ID number.

If mode = 1, the object is written at the first available ID number after what is specified by idno.

It doesn't matter if idno is a legal ID number. Remember that for certain object types, the 100s through 900s banks allow fewer than 100 objects to be stored (for example, the 100s bank will store Quick-Access banks at IDs 100–119 only). In this mode, if idno were 313, the object would be written to ID 400 if available.

READ = 0Ah type(2) idno(2) form(1)

Requests the K2600 to send a WRITE message for the given object. No response will be sent if the object does not exist.

READBANK = 0Bh type(2) bank(1) form(1) ramonly(1)

Requests the K2600 to send a WRITE message for multiple objects within one or all banks.

type and bank specify the group of objects to be returned in WRITE messages. The type field specifies a single object type, unless it is zero, in which case objects of all user types will be returned (see object type table below). The bank field specifies a single bank, 0–9, unless it is set to 127, in which case objects from all banks will be returned.

form requests the format of the binary data in the WRITE messages. If ramonly is one, only objects in RAM will be returned. If ramonly is zero, both RAM and ROM objects are returned.

The responses, a stream of complete WRITE messages, will come out in order of object type, while objects of a given type are in order by ID number, from lowest to highest. If no objects are found that match the specifications, no WRITE messages will be returned. After the last WRITE message, an ENDOFBANK message (defined below) is sent to indicate the completion of the bank dump.

The K2600 will insert a small delay (50ms) between WRITE messages that it issues in response to a READBANK message.

A bank dump can be sent in its entirety to another K2600, which will add all of the objects contained in the dump to its own object database. Important: If the K2600 receives a large bank dump for a bank or banks that already contain objects, errors may result unless the sender waits for the DACK message before sending the next object's WRITE message. One way to avoid transmission errors such as this is to make sure that the bank being dumped is clear in the K2600 before sending the dump, so that the K2600 will not miss parts of the dump while its CPU is busy deleting already existing objects. This can be done using the DELBANK message (defined below). If the destination bank in the K2600 is clear, it is not necessary to wait for the DACK before sending. Even if the sender chooses not to wait for the DACK before sending the next message, it may be necessary to preserve the 50ms delay between the WRITE messages.

Due to the large amount of incoming data during a bank dump containing many objects, the receiving K2600 may have a more sluggish response to front-panel use and keyboard playing during the data transfer. This is normal behavior and the machine will become fully responsive as soon as the dump is finished.

DIRBANK = 0Ch type(2) bank(1) ramonly(1)

This is similar to the READBANK message. The DIRBANK message requests an INFO message (containing object size, name, and memory information) be returned for each object meeting the specifications in the type and bank fields. Following the last INFO response will be an ENDOFBANK message.

ENDOFBANK = 0Dh type(2) bank(1)

This message is returned after the last WRITE or INFO response to a READBANK or DIRBANK message. If no objects matched the specifications in one of these messages, ENDOFBANK will be the only response.

DELBANK = 0Eh type(2) bank(1)

This message will cause banks of objects (of one or all types) to be deleted from RAM. The type and bank specifications are the same as for the READBANK message. The deletion will take place with no confirmation. Specifically, the sender of this message could just as easily delete every RAM object from the K2600 (for example, type = 0 and bank = 127) as it could delete all studios from bank 7 (for example, type = 113, bank = 7.)

MOVEBANK = 0Fh type(2) bank(1) newbank(1)

This message is used to move entire banks of RAM objects from one bank to another. A specific object type may be selected with the type field. Otherwise, if the type field is unspecified (0), all object types in the bank will be moved. The bank and newbank fields must be between 0 and 9. The acknowledgement is an ENDOFBANK message, with the bank field equal to the new bank number. If the operation can't be completed because of a bad type or bank number, the ENDOFBANK message will specify the old bank number.

LOADMACRO = 10h

Tells K2600 to load in the macro currently in memory.

MACRODONE = 11h code(1)

Acknowledges loading of macro. Code 0 indicates success; code 1 means failure.

PANEL = 14h buttons(3n)

Sends a sequence of front-panel button presses that are interpreted by the K2600 as if the buttons were pressed at its front panel. The button codes are listed in tables beginning on page 7-7. The K2600 will send these messages if the Buttons parameter on the TRANSMIT page in MIDI mode is set to **On**. Each button press is 3 bytes in the message. The PANEL message can include as many 3-byte segments as necessary.

Byte 1	Button event type:
08h	Button up
09h	Button down
0Ah	Button repeat
0Dh	Alpha Wheel
Byte 2	Button number (see table)
Byte 3	Repeat count (number of clicks) for Alpha Wheel; the count is the delta (difference) from 64—that is, the value of the byte minus 64 equals the number of clicks. A Byte 3 value of 46h (70 dec) equates to 6 clicks to the right. A Byte 3 value of 3Ah (58 dec) equates to six clicks to the left. For example, the equivalent of 6 clicks to the right would be the following message: (header) 14h 0Dh 40h 46 (eox)

For efficiency, multiple button presses should be handled by sending multiple Button down bytes followed by a single Button up byte (for incrementing with the **Plus** button, for instance).

Object Types

These are the object types and the values that represent them in type fields:

Type	ID (decimal)	ID (hex)	ID (hex, type field)
Program	132	84h	01h 04h
Keymap	133	85h	01h 05h
Studio	113	71h	00h 71h
Song	112	70h	00h 70h
Setup	135	87h	01h 07h
Soundblock	134	86h	01h 06h
Velocity Map	104	68h	00h 68h
Pressure Map	105	69h	00h 69h
Quick Access Bank	111	6Fh	00h 6Fh
Intonation Table	103	67h	00h 67h

Master Parameters

The Master parameters can be accessed as type 100 (00h 64h), ID number 16. Master parameters cannot be accessed with any of the Bank messages.

Button Press Equivalence Tables

Alphanumeric pad		Soft-Buttons A-F	
Button	**Code (hex)**	**Button**	**Code(hex)**
zero	00	A (leftmost)	22
one	01	B	23
two	02	C	24
three	03	D	25
four	04	E	26
five	05	F (rightmost)	27
six	06	AB	28
seven	07	CD (two center)	29
eight	08	EF	2A
nine	09	YES	26
+/-	0A	NO	27

Alphanumeric pad		Edit / Exit	
Button	**Code (hex)**	**Button**	**Code(hex)**
Cancel	0B	Edit	20
Clear	0C	Exit	21
Enter	0D		

Navigation		Mode Selection	
Button	**Code (hex)**	**Button**	**Code(hex)**
Plus (+)	16	Program	40
Minus (-)	17	Setup	41
Plus and Minus	1E	Quick Access	42
Chan/Bank Inc	14	Effects	47
Chan/Bank Dec	15	Midi	44
Chan/Bank Inc/Dec	1C	Master	43
Cursor Left	12	Song	46
Cursor Right	13	Disk	45
Cursor Left/Right	1A		
Cursor Up	10		
Cursor Down	11		
Cursor Up/Down	18		

The next four commands allow you to read the screen display, both text and graphics layers.

ALLTEXT = 15h

…requests all text in the K2600’s display.

PARAMVALUE = 16h

…requests the current parameter value.

PARAMNAME = 17h

…requests the current parameter name.

GETGRAPHICS = 18h

…requests the current graphics layer.

SCREENREPLY = 19h

This is the reply to ALLTEXT, PARAMVAL, PARAMNAME, GETGRAPHICS, or SCREENREPLY.

The reply to ALLTEXT will be 320 bytes of ASCII text (the display has 8 rows of 40 characters each). If you receive less than that, then the screen was in the middle of redrawing and you should request the display again.

The reply to PARAMVALUE will be a variable length ASCII text string. Some values (like keymaps, programs, samples, etc.) include their ID number in the text string (for example, **983 OB Wave 1**). Some messages are also padded with extra spaces.

The reply to PARAMNAME will be a variable length ASCII text string. In cases where there is no parameter name (like on the program page) there will just be the single 00 null terminator.

The reply to GETGRAPHICS will be 2560 bytes of information. The 6 least significant bits of each byte indicate whether a pixel is on or off. If pixels are on over characters, the text becomes inverted. Characters on the K2600 display are a monospaced font with a height of 8 pixels and a width of 6 pixels.

Chapter 8
Maintenance and Troubleshooting

Preventive Maintenance

With a modicum of care, your K2600 will give you years of use and enjoyment. There are just a few important points to keep in mind.

Proper installation is essential to the health and welfare of your K2600. Keyboard models should always rest on a hard flat surface—and on its rubber feet, not on the bottom panel. Rack models should be mounted in a standard 19-inch MIDI rack, or should rest on a hard flat surface. In this case it must rest on its rubber feet, and not on the bottom panel.

Never block the ventilation openings; doing so can cause overheating that will seriously damage your K2600. To provide adequate ventilation, the rear panel should be at least four inches from any vertical surface (for both keyboard and rack models). Try to minimize the amount of dust in the environment.

The K2600's RAM backup battery, along with any PRAM, sample RAM, or ROM block options you may install, are the only user-serviceable parts in the K2600. The only part you should ever disassemble on your K2600 is the access panel on the bottom of the instrument—removing anything else will void your product warranty.

Cleaning Your K2600

It's a good idea to remove dust from your K2600 occasionally. You may also want to remove fingerprints. You can clean the K2600's front panel with a soft damp cloth, and use a mild soap or detergent. Never use strong cleaners or solvents, and never spray anything on the front panel or into the ventilation holes. Any cleaners you may want to use should be applied to your cleaning cloth; you can then carefully wipe the surfaces of the K2600.

Floppy Disk Drive Maintenance

As long as you're reasonably careful to keep dirt and dust out of the floppy disk drive, you shouldn't have any problems. If, however, you start to experience errors or failures in loading or saving, it may be due to dirt in the floppy drive mechanism. See your dealer for information regarding products and techniques for floppy drive cleaning.

Battery Replacement

The K2600 uses a 3-volt lithium coin-cell battery (CR2032) for program RAM backup (sample RAM is not battery-backed). Unlike a typical alkaline battery—whose voltage output declines over the life of the battery—a lithium cell maintains a stable voltage until it's almost out of power. Once it has used up almost all of its power, however, its voltage drops rapidly. Consequently, to avoid the risk of losing the contents of your program RAM, you should replace the battery as soon as your K2600 warns of low battery voltage.

The battery in your K2600 will last for several years (fewer if you add the P/RAM-26 option, which approximately doubles the load on the battery).You'll know the battery is losing power when the display says BATTERY VOLTAGE IS LOW during powerup. When you see this warning, replace the battery as soon as possible.

Replacing the battery requires you to open an access panel on the underside of your instrument. This is the same panel you would open to install program RAM (P/RAM-26), sample RAM, or ROM sound block options.

1. Obtain a CR2032 lithium coin cell; any store that sells batteries for small electronic appliances is likely to have them in stock.
2. Make sure you have backups of any RAM objects (not including samples) in the K2600 that you really care about. A quick way to make backups is to use the save function in Disk mode, and choose to save everything instead of choosing one bank at a time.

Warning*: Turn off your K2600 and disconnect the power cable!*

3. Carefully place your K2600 upside down on a padded level surface, with the front of the instrument toward you. Keyboard owners should use soft, sturdy foam under the ends of the instrument, to protect the wheels and sliders.
4. Locate the access panel. On keyboard models, it's about 6 by 13 inches in size, slightly to the right of center, toward the back of the instrument. On rack models, it's about 7 by 12 inches, and takes up a large portion of the underside of the instrument.
5. Remove the screws that hold the access panel in place—eight for the keyboard and six for the rack—and swing the panel open from the front. It hinges at the back, and rests in a position that's convenient for referring to the diagram that's printed on the inside of the panel.
6. Locate the battery slot. It's toward the far edge of the circuit board, toward the rear of the instrument.
7. Put the new battery in an easily-accessible location. Once you remove the old battery, you'll have about 30 seconds to install the new one before you lose data from program RAM. If you install the new battery within 30 seconds, you probably won't have to reload any program-RAM objects.
8. To snap the old battery out of its retaining clip, lift up on the front of the battery (there's a notch at the front of the clip, where you can get a bit of leverage), then push the battery toward you from behind. If necessary, carefully use a small screwdriver or other object to push the battery out.
9. Snap the new battery in place, with the plus side up. Make sure that it snaps securely into the retaining clip.

10. Replace the access panel and loosely install the screws, starting with those closest to the hinge (the back) of the access panel. When the screws are loosely in place, tighten them all.

***Note to K2500 owners**: On 2500-series instruments, the LEDs on the front panel flash three times when battery voltage is low. This isn't necessarily the case with the K2600—in fact on rack models, the LEDs always flash three times at powerup.*

Scanner Diagnostics

There's an onboard diagnostic program that enables you to check your battery and confirm front panel button functions.

To enter the Scanner Diagnostics, simply press **4**, **5**, and **6** (on the alphanumeric buttonpad) simultaneously while in Program mode. The K2600 responds by lighting each LED in sequence and then displaying something like the following (the display says K2500, but the diagnostics have been updated for the K2600).

```
K2600 SCANNER DIAGNOSTICS VERSION 5.00
(PRESS "EXIT" AND "ENTER" TO EXIT)
BATTERY=3.2VOLTS, WHEEL CENTER=128

XXXXXX
```

The battery voltage and wheel center values may be different on your unit. The line represented by XXXX gives a readout identifying the buttons you press.

The diagnostic program can also be used to check out the front panel components. If you move the Alpha Wheel clockwise, the numbers will go 0-1-2-3-0-1-2...while counterclockwise should produce 3-2-1-0-3-2-1... If you press a button, its name will be shown and if it is one of the mode buttons, its associated LED should flash.

The third line of the display shows the results of two measurements that are made whenever your K2600 is turned on. The battery voltage will be about 4.3 volts for new batteries, gradually declining over time to 3.2 volts, at which point you will begin to receive warnings (see *Battery Replacement* on page 8-2). The line referring to the Wheel is relevant for keyboard models only.

Maximizing Music and Minimizing Noise

Your K2600 quite possibly has the lowest noise and widest dynamic range of any instrument in your studio. The following tips will enable you to make the most of this, and optimize the K2600's audio interface to your other equipment.

Setting your audio levels appropriately is the key to optimizing the signal-to-noise ratio of any piece of equipment. It's best to increase the output level *digitally* (by editing programs) instead of increasing the gain of your amplifier or mixing board. This is because a digital gain increase is completely noiseless, whereas an analog increase will proportionally increase hum and noise from the connecting cabling and from the K2600 itself.

Increasing the volume digitally can be accomplished in three different ways. You can increase the volume of all programs assigned to a given MIDI channel by selecting the CHANNELS page in MIDI mode and setting the OutGain parameter to the desired level (in 6dB steps). For songs that use multiple MIDI channels, you'll need to do this for each channel. Alternatively you can increase the volume of a single program by going to the OUTPUT page in the Program Editor and setting the Gain parameter to the desired level, again in 6dB steps. For finer adjustment, there's the Adjust parameter on the F4 AMP page.

Increasing the level too much can cause clipping distortion when multiple notes are triggered with high attack velocity. For dense songs played through the same outputs, you will probably be able to increase the volume by only 6 dB or so without risk of distortion. For monophonic instruments (lead guitar) or single instrument tracks (such as drums), a substantially greater boost is generally possible.

For the absolute maximum signal quality (with the exception of digital output, of course), use the separate analog outputs. These are connected almost directly to the 18-bit digital-to-analog converters with a minimum of noise-inducing processing circuitry. A total dynamic range of over 100dB is available at these outputs. The MIX outputs are naturally somewhat noisier because they represent the noise of the individual outputs mixed together, and the signal must travel through more circuitry to reach them.

Ground Hum

A common problem with all electrical musical gear is the hum that can occur in connecting cables due to AC ground loops. The best way to avoid ground loop noise when integrating the K2600 into a stage or studio environment is to use the K2600's balanced audio outputs, and to be sure that the mixing board, amplifier, or other equipment receiving the K2600 audio signal has a balanced input circuit.

If you can't use the K2600 audio outputs in a balanced manner, there are a few things you can do to reduce ground hum. Although "3-prong to 2-prong" AC adapters are frequently used to break ground loops, they also break the safety ground that protects you from electric shock. These adapters can be dangerous; don't use them! Furthermore, although using these adapters may reduce low-frequency hum, high-frequency line noise (such as motor switching noise) is likely to get worse in this case, since the K2600's AC noise filter will have no output for the noise it filters if you disable the ground.

You can effectively reduce hum by increasing your output signal levels as described in the previous section. Other safe procedures include plugging your mixing board and amplifier into the same AC output as your K2600, and making sure that all of your gear is properly grounded. If you're using an external SCSI device, plug it into the same outlet as well.

AC isolation transformers are extremely effective at eliminating ground loops, and are recommended for critical installations in which you can't use the K2600's balanced outputs. A 75-watt transformer is sufficient for the K2600.

Use the shortest possible cable, with the heaviest possible ground (shield) wire, to connect your K2600 to the mixing board or amplifier. This helps to reduce the potential difference between the chassis of the K2600 and the chassis of a mixing board or amplifier that has unbalanced inputs—thus reducing the level of ground hum.

Finally, magnetic fields can be a source of interference. The area surrounding the K2600's Alpha Wheel and alphanumeric buttonpad is sensitive to fields from large transformers in power amps; keep them at least a foot away from the K2600's front panel. Smaller gear like drum machines and hardware sequencers can also cause interference.

Power Problems and Solutions

The K2600 is quite tolerant of voltage fluctuations, noise, and transients in the AC power it receives. The input line filter and grounded power cable will protect against even large amounts of noise from motors and the like while the built-in filter coupled with the fuse will protect against all but the largest transients. If your installation is actually suffering from line noise or transients, most likely your other equipment will be suffering more than your K2600.

Very low line voltage or severe voltage dips are a problem for any computer-based instrument. When the K2600 is set for 120 volt input (the normal North American setting), it should function down to 90 volts. If the line voltage drops below 90 volts, a special circuit halts all activity to protect against software crashes or damage. When the line voltage returns to and stays at an acceptable level for at least one second, the computer will automatically restart. The net effect is just as if you had performed a soft reset. Continuous low line voltage or transient dips will never produce symptoms other than unexpected soft resets as just described. Any other problems such as distortion, disk errors, or lost data are caused by something other than line voltage fluctuations.

Soft resets from line voltage dips are most common. These are easily identified because the reset occurs coincident with the building lights dimming, stage lights or power amps being switched on, or air-conditioning equipment starting up. The solution in all cases is to get a more direct connection between your K2600 (and any other computer-based equipment) and the building's power. Floodlights, large power amplifiers, and motor-operated devices should use a separate extension cord; preferably they should be plugged into a separate circuit.

Chronic low line voltage is best confirmed by measurement. Readings below 100-105 volts mean that even small dips could cause resets, while readings below 95 volts (accounting for meter inaccuracies) are a definite problem. Again, the best solution is to separate your heavy lighting and amplifier loads from your K2600 and other synths on separate extension cords or separate circuits when possible. If the actual building voltage is that low, we recommend using an external step-up transformer or voltage regulator. We *do not* recommend changing the line voltage selector to 100 volts (or 220 volts in Europe) because overheating or blown fuses may occur if you leave the K2600 at the lower setting and use it later at a normal voltage level.

Troubleshooting

Naturally, we've done everything possible to ensure that your K2600 arrives free of defects. And there's a good chance that there's nothing wrong, even if you're not seeing the proper display or hearing the sounds. Carefully check the following things:

Make sure that your power supply is at the right voltage, and is functioning properly.

Make sure the power cable is connected properly.

Adjust the display contrast if necessary (on keyboard models, there's a knob on the rear panel; on rack models, the knob is on the front panel, above the power switch). If you still have trouble seeing the display, it's time to contact your dealer.

Make sure your audio cables are fully connected to the K2600 and to your sound system. You may want to switch your audio cables, unless you're sure they're functioning properly.

For rack models, make sure that your MIDI connections are correct, and that your MIDI cables are functional. You should have at least one MIDI cable, which should be connected from the MIDI Out port of your MIDI source to the MIDI In port of the K2600.

Check that the K2600's Volume slider is at least partially up.

Check the volume level of your sound system.

Lower the volume of your sound system, and turn the K2600 off, then on again (this is called a power cycle).

Press the **+/-**, **0**, and **Clear** buttons (on the alphanumeric buttonpad) at the same time. This is called a soft reset.

As a last resort, save to disk any RAM objects you've created, and perform a hard reset. Do this by pressing the Master-mode button, then pressing the **Reset** soft button (at the lower right of the display). The K2600 will warn you about deleting everything (only RAM objects will be deleted). Press **Yes**. After a few seconds, the power-up display should appear.

Also check the suggestions on the following pages. If it's still not happening, the next step is to shut off the power and call your dealer.

Other Possible Problems

No Sound, No Display, No LEDs Illuminated

1. AC line cord not fully inserted into outlet or unit. If using a multiple outlet box, check its plug.
2. Power not on at AC power source (wall outlet). Check with a different appliance.
3. Power switch not on (either the unit or multiple outlet box).
4. Incorrect voltage selection setting. REFER TO QUALIFIED SERVICE PERSONNEL.

No Sound

1. Volume control turned all the way down on the K2600 or on amplifier or mixer.
2. Amplifier or mixer not turned on.
3. Cabling is not correct; see Chapter 2 of the*Musician's Guide*, and read about the various cable connections you need to make: power, audio, MIDI. There's more about audio configurations in Chapter 19 of the *Musician's Guide*. Also check that your amplifier, mixer and speaker connections are correct.
4. MIDI volume has been assigned to a control source which has sent a value of 0. Pressing the **Panic** soft button will reset all controls, and resolve this problem.

Left MIX Output Seems Louder Than Right MIX Output When Used Individually

This is normal. When a cable is plugged into the left MIX output alone, both the left AND the right audio signals are routed to the jack. When a cable is plugged into the right MIX output alone, only the right channel audio signal is heard.

Volume Knob Has No Effect

1. Separate outputs are in use; the volume knob does not affect the separate outputs.
2. MIDI volume has been assigned to a control source which has sent a value of 0.

Programs, Setups, Songs, or Other Objects Are Missing

Battery has run down or has been disconnected. If the battery has failed, the message "Battery voltage is low - X.X volts" (where X.X is less than 3.0) will appear in the display on powerup. All user data will be permanently lost if this occurs. See *Battery Replacement* on page 8-2.

Cannot Mount or Read Disk

1. Disk is not MS-DOS (or Akai, Ensoniq, or Roland) format.
2. Disk is damaged.

Cannot Write to Floppy Disk

1. Disk is not MS-DOS formatted.
2. Disk is write protected.
3. Sample is copy protected.
4. Disk is damaged.

Cannot Format Disk

1. Disk is damaged.
2. Disk is write protected.
3. You have instructed the K2600 to format a double-density (720K) disk as a high-density (1.4M) disk. Note: Punching a hole in a double-density disk case to try to make the K2600 read it as a high-density disk is not a good idea.

Chapter 9
Memory Upgrades and Other Options

Program RAM vs. Sample RAM

If you’re creating a lot of your own programs, and using samples loaded from disk, there are a few things you should be aware of to avoid perplexity. First of all, there’s an important distinction between what we call sample RAM and what we call program RAM. Sample RAM refers to any SIMMs you may have had installed in your K2600. This RAM is reserved exclusively for sample storage; nothing else is stored there. Sample RAM is volatile; that is, when you power down your K2600, the data stored there will “evaporate” almost immediately. That’s why you have to load RAM samples every time you power up.

The amount of sample RAM in your K2600 is indicated in the center of the top line of the Disk-mode page. If the center of the display’s top line is blank when you’re on this page, it means that there is no sample RAM installed in your K2600 (or that the K2600 isn’t recognizing it, in which case you should see your dealer or service center).

Program RAM is where all the other RAM objects you create (programs, setups, QA banks, songs, keymaps, etc.) are stored. The K2600 comes from the factory with approximately 500K of available program RAM. The amount of free program RAM is indicated at the right side of the top line of the display in Song mode and Disk mode. You can add a program RAM (P/RAM) option to increase your total available program RAM to about 1500K. Ask your dealer.

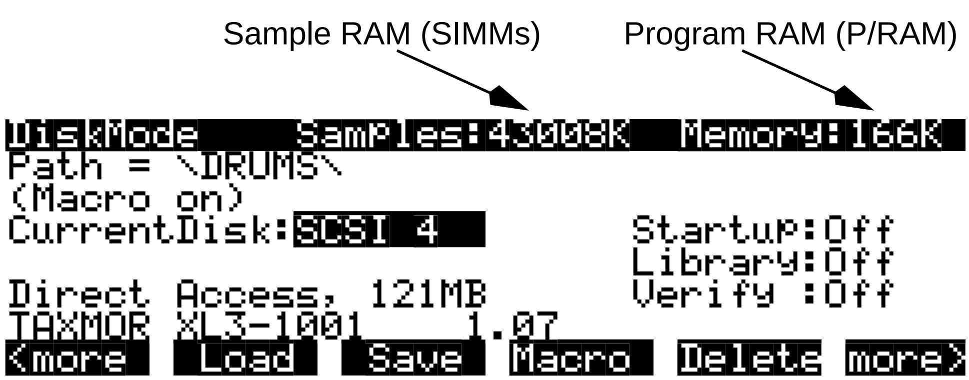


Figure 9-1 Disk mode page showing Sample RAM and Program RAM

Program RAM is battery-backed, so anything that’s stored there will be preserved even when you power down (as long as your battery is functional). A fresh lithium battery should last for several years, so you’ll have very few worries about losing your RAM program information. Nonetheless, we recommend that you back up programs, songs, etc. by saving them to disk. This offers insurance in case the RAM becomes corrupted. This is unlikely, but still a possibility.

If you create a program that uses a disk-loaded sample, the program information (number of layers, keymap assignment, output group, algorithm, etc.) is stored in program RAM. All RAM samples associated with the program are stored in sample RAM. This means that when you power down, the RAM samples associated with your programs will disappear. The program information, however, will remain in program RAM indefinitely. When you power up again, your RAM programs will still appear in the display as you scroll through the program list, but they won't play if they use RAM samples, because the RAM samples are lost when you power down.

Viewing RAM Objects

If you're a heavy Disk-mode user, you'll often be faced with the decision to overwrite, merge, or append objects when you load files from disk. If you're loading into a memory bank that's nearly full, this can be a tricky call, because if you decide to merge or append, there may not be enough open slots in the memory bank to accommodate the objects you load. In this case, the extra objects will be loaded into the next-higher memory bank.

Things get even trickier if you save dependent objects when you save to disk. (A dependent object is any object that's associated with another object stored in a different memory bank—for example, a RAM sample with ID 301 that's used in a program with ID 200. See the discussions of dependent objects on page 13-18 and page 13-29 of the *Musician's Guide*. If you load a file that contains a number of dependent objects, some of them may be loaded into a higher memory bank than the one you specified in the Bank dialog before you loaded the file. A quick way to see where the objects you loaded ended up is to use the Objects utility function in Master mode.

Select Master mode and press the **Utility** soft button. Press the **Objects** soft button, and a list of RAM objects will appear. Use the Alpha Wheel to scroll through the list of objects. You'll see the type, ID, name, and size (in bytes) of each object.

Choosing and Installing SIMMs for K2600 Sample Memory

SIMM Specifications

SIMMs for sample RAM must have the following characteristics:

- 72-pin noncomposite single, in-line memory modules (SIMMs), in sizes of 4 M, 8 M, 16 M, 32 M, 64 M, or 128 M
- 8- or 9-bit
- 3-volt or 5-volt (most SIMMS currently on the market are 5-volt)
- Fast-page (FPM) or extra data output (EDO) (80-nanosecond or faster)

You can add one or two SIMMs, up to a total of 128 M. See Table 9-1 on page 9-3 for size compatibility requrements.

SIMM Configurations

Some SIMMs cannot be paired with other SIMMs, regardless of their sizes. The following table shows which sizes can be combined.

Size in Megabytes	Can Be Paired With Other SIMMs
4	Yes
8	No
16	Yes
32	No
64	Yes
128	No

Table 9-1 SIMM-size Compatibility

For example, a 4 M SIMM can be combined with another 4 M SIMM to create 8 M of sample memory. Similarly, a 4 M SIMM could be paired with a 16 M or 64 M SIMM. It could not, however, be paired with an 8 M, 32 M or 128 M SIMM. If you use an 8 M, 32 M, or 128 M SIMM, you cannot use the other SIMM socket.

These companies make SIMMs that work (many other sources are also likely to have the proper configurations):

Newer RAM	(800) 678-3726 or (316) 943-0222
Chip Merchant	(800) 808-2447 or (619) 268-4774
Kamel Peripherals	(508) 435-7771 or (888) 295-2635
Lifetime Memory	(800) 233 6233 or (714) 794-9000

***Caution**: Do not use composite SIMMs. A composite SIMM is one that uses a PAL or other additional circuitry to make multiple DRAM chips act like bigger chips. Non-composite SIMMs (acceptable) have no chips other than DRAM memory chips soldered to the board. SIMMs with PALs, buffers, or other logic components will not work in a K2600; do not use them. Composite SIMMs may appear to work in some cases, but they will be unreliable.*

Installing Sample RAM

There's an access panel on the underside of your K2600, which you'll need to open to install your sample RAM. This is the same panel you would open to install a replacement battery, the P/RAM-26 option, or ROM sound block options.

***Warning**: Turn off your K2600 and disconnect the power cable!*

1. Carefully place your K2600 upside down on a padded level surface, with the front of the instrument toward you. Keyboard owners should use soft, sturdy foam under the ends of the instrument, to protect the wheels and sliders.

2. Locate the access panel. On keyboard models, it's about 6 by 13 inches in size, slightly to the right of center, toward the back of the instrument. On rack models, it's about 7 by 12 inches, and takes up a large portion of the underside of the instrument.

3. Remove the screws that hold the access panel in place—eight for the keyboard and six for the rack—and swing the panel open from the front. It hinges at the back, and rests in a position that's convenient for referring to the diagram that's printed on the inside of the panel.

4. Locate the two sockets for sample RAM. Note the location of the socket for the P/RAM-26 option. Don't put sample RAM into the P/RAM socket! Doing so could cause serious damage to the K2600, the SIMM, or both.

5. Place a SIMM into one or both of the sample RAM sockets. If you're putting in two SIMMs, be sure that their sizes are compatible, as shown in Table 9-1. There's only one way that the SIMMs will fit into the sockets; they won't fit at all if they're facing the wrong way. Be sure the clips at the sides of the sockets snap into place.

6. Check the setting of the voltage jumper, and change it if it doesn't match the voltage of your SIMMs. The K2600 arrives from the factory with the jumper set for 3-volt SIMMs. Since most SIMMs these days are 5-volt, you'll probably need to change the jumper setting.

   The jumper is a small piece of molded plastic with a wire loop at the top. It has two slots that slide over two of the three pins that stick up from the circuit board. The pins are numbered from 1 to 3, *right-to-left*. Put the jumper on pins 2 and 1 (the two right-most pins) to configure the K2600 for 5-volt SIMMs, or on pins 3 and 2 (the two left-most pins) to configure it for 3-volt SIMMS. The circuit board is labeled accordingly.

   We set the configuration for 3 volts so that if you were to install 3-volt SIMMs while thinking you were installing 5-volt SIMMS, you wouldn't pose any risk to your instrument.

7. Replace the access panel and loosely install the screws, starting with those closest to the hinge (the back) of the access panel. When the screws are loosely in place, tighten them all.

Using Headphones with the K2600

A good pair of headphones can be indispensable when you want to play but need to keep the volume down. You'll get optimum performance from headphones with at least 50 ohms impedance, but anything over eight ohms is adequate. Headphone volume decreases as the impedance decreases.

Chapter 10
KDFX Reference

In This Chapter

KDFX Algorithms

Reverb Algorithms

ID	Name
1	MiniVerb
2	Dual MiniVerb
3	Gated MiniVerb
4	Classic Place
5	Classic Verb
6	TQ Place
7	TQ Verb
8	Diffuse Place
9	Diffuse Verb
10	OmniPlace
11	OmniVerb
12	Panaural Room
13	Stereo Hall
14	Grand Plate
15	Finite Verb

Delay Algorithms

ID	Name
130	Complex Echo
131	4-Tap Delay
132	4-Tap Delay BPM
133	8-Tap Delay
134	8-Tap Delay BPM
135	Spectral 4-Tap
136	Spectral 6-Tap

Chorus / Flange / Phaser Algorithms

ID	Name
150	Chorus 1
151	Chorus 2
152	Dual Chorus 1
153	Dual Chorus 2
154	Flanger 1
155	Flanger 2
156	LFO Phaser
157	LFO Phaser Twin
158	Manual Phaser
159	Vibrato Phaser
160	SingleLFO Phaser

Combination Algorithms

ID	Name
700	Chorus+Delay
701	Chorus+4Tap
702	Chorus<>4Tap
703	Chor+Dly+Reverb
704	Chorus<>Reverb
705	Chorus<>LasrDly
706	Flange+Delay
707	Flange+4Tap
708	Flange<>4Tap
709	Flan+Dly+Reverb
710	Flange<>Reverb
711	Flange<>LasrDly
712	Flange<>Pitcher
713	Flange<>Shaper
714	Quantize+Flange
715	Dual MovDelay
716	Quad MovDelay
717	LasrDly<>Reverb
718	Shaper<>Reverb
719	Reverb<>Compress
720	MonoPitcher+Chor
721	MonoPitcher+Flan
722	Pitcher+Chor+Dly
723	Pitcher+Flan+Dly

Distortion Algorithms

ID	Name
724	Mono Distortion
725	MonoDistort+Cab
726	MonoDistort + EQ
727	PolyDistort + EQ
728	StereoDistort+EQ
729	TubeAmp<>MD>Chor
730	TubeAmp<>MD>Flan
731	PolyAmp<>MD>Chor
732	PolyAmp<>MD>Flan

Tone Wheel Organ Algorithms

ID	Name
733	VibChor+Rotor 2
734	Distort + Rotary
735	KB3 FXBus
736	KB3 AuxFX
737	VibChor+Rotor 4

Special FX Algorithms

ID	Name
900	Env Follow Filt
901	TrigEnvelopeFilt
902	LFO Sweep Filter
903	Resonant Filter
904	Dual Res Filter
905	EQ Morpher
906	Mono EQ Morpher
907	Ring Modulator
908	Pitcher
909	Super Shaper
910	3 Band Shaper
911	Mono LaserVerb
912	LaserVerb Lite
913	LaserVerb

Studio / Mixdown FX Algorithms

ID	Name
950	HardKneeCompress
951	SoftKneeCompress
952	Expander
953	Compress w/SC EQ
954	Compress/Expand
955	Comp/Exp + EQ
956	Compress 3 Band
957	Gate
958	Super Gate
959	2 Band Enhancer
960	3 Band Enhancer
961	Tremolo
962	Tremolo BPM
963	AutoPanner
964	Dual AutoPanner
965	SRS
966	Stereo Image
967	Mono -> Stereo
968	Graphic EQ
969	Dual Graphic EQ
970	5 Band EQ

Tools

ID	Name
998	FXMod Diagnostic
999	Stereo Analyze

KDFX Presets

ID	Preset Name	KDFX Alg
1	NiceLittleBooth	1
2	Small Wood Booth	4
3	Natural Room	5
4	PrettySmallPlace	4
5	Sun Room	5
6	Soundboard	7
7	Add More Air	10
8	Standard Booth	8
9	A Distance Away	6
10	Live Place	8
15	BrightSmallRoom	1
16	Bassy Room	1
17	Percussive Room	1
18	SmallStudioRoom	4
19	ClassRoom	5
20	Utility Room	5
21	Thick Room	5
22	The Real Room	5
23	Sizzly Drum Room	5
24	Real Big Room	5
25	The Comfy Club	9
26	Spitty Drum Room	7
27	Stall One	7
28	Green Room	7
29	Tabla Room	12
30	Large Room	7
31	Platey Room	14
40	SmallDrumChamber	1
41	Brass Chamber	1
42	Sax Chamber	1
43	Plebe Chamber	1
44	In The Studio	4
45	My Garage	4
46	School Stairwell	4
47	JudgeJudyChamber	7
48	Bloom Chamber	7
55	Grandiose Hall	1
56	Elegant Hall	1
57	Bright Hall	1
58	Ballroom	1
59	Spacious Hall	5
60	Classic Chapel	5
61	Semisweet Hall	5
62	Pipes Hall	704
63	Reflective Hall	5
64	Smoooth Hall	5
65	Splendid Palace	5
66	Pad Space	11
67	Bob'sDiffuseHall	9
68	Abbey Piano Hall	7
69	Short Hall	13
70	The Long Haul	7
71	Predelay Hall	9
72	Sweeter Hall	7
73	The Piano Hall	7
74	Bloom Hall	9
75	Recital Hall	12
76	Generic Hall	12
77	Burst Space	9
78	Real Dense Hall	7
79	Concert Hall	9
80	Standing Ovation	11
81	Flinty Hall	7
82	HighSchool Gym	7
83	My Dreamy 481!!	9
84	Deep Hall	9
85	Immense Mosque	7
86	Dreamverb	10
87	Huge Batcave	12
95	Classic Plate	5
96	Weighty Platey	5
97	Medm Warm Plate	7
98	Bloom Plate	9
99	Clean Plate	9
100	Plate Mail	11
101	RealSmoothPlate	9
102	Huge Tight Plate	9
103	BigPredelayPlate	7
110	L:SmlRm R:LrgRm	2
111	L:SmlRm R:Hall	2
112	Gated Reverb	3
113	Gate Plate	3
114	Exponent Booth	10
115	Drum Latch1	10
116	Drum Latch2	10
117	Diffuse Gate	9
118	Acid Trip Room	10
119	Furbelows	9
120	Festoons	9
121	Reverse Reverb	15
130	Guitar Echo	130
131	Stereo Echoes1	130
132	Stereo Echoes2	130
133	4-Tap Delay	132
134	OffbeatFlamDelay	132
135	8-Tap Delay	134
136	Spectral 4-Tap	135
137	Astral Taps	135
138	SpectraShapeTaps	136
150	Basic Chorus	152
151	Chorus Comeback	152
152	Chorusier	152
153	Ordinary Chorus	152
154	SlowSpinChorus	152
155	Chorus Morris	152
156	Everyday Chorus	152
157	Thick Chorus	153
158	Soft Chorus	153
159	Rock Chorus	153
160	Sm Stereo Chorus	150
161	Lg Stereo Chorus	151
170	Big Slow Flange	154
171	Wetlip Flange	154
172	Sweet Flange	154
173	Throaty Flange	154
174	Delirium Tremens	154
175	Flanger Double	154
176	Squeeze Flange	154
177	Simply Flange	155
178	Analog Flanger	155
190	Circles	156
191	Slow Deep Phaser	157
192	Manual Phaser	158
193	Vibrato Phaser	159
194	ThunderPhaser	159
195	Saucepan Phaser	160
199	No Effect	0
700	Chorus Delay	700
701	Chorus PanDelay	700
702	Doubler & Echo	700
703	Chorus VryLngDly	700
704	FastChorusDouble	700
705	BasicChorusDelay	700
706	MultiTap Chorus	701
707	ThickChorus no4T	701
708	Chorused Taps	702
709	Chorus Slapbacks	705
710	MultiEchoChorus	705
711	ChorusDelayHall	703
712	ChorDlyRvb Lead	703
713	ChorDlyRvb Lead2	703
714	Fluid ChorDlyRvb	703
715	ChorLite DlyHall	703
716	ChorusSmallRoom	703
717	DeepChorDlyHall	703
718	Chorus PercHall	703
719	Chorus Booth	703
720	ClassicEP ChorRm	703

ID	Preset Name	KDFX Alg
721	ChorusMedChamber	704
722	Vanilla ChorRvb	704
723	Chorus Slow Hall	704
724	SoftChorus Hall	704
725	ChorBigBrtPlate	704
726	Chorus Air	704
727	Chorus HiCeiling	704
728	Chorus MiniHall	704
729	CathedralChorus	704
730	PsiloChorusHall	704
731	GuitarChorLsrDly	705
732	Flange + Delay	706
733	ThroatyFlangeDly	706
734	Flange + 4Tap	707
735	Bap ba-da-dap	707
736	Slapback Flange	706
737	Quantize+Flange	714
738	FlangeDelayHall	709
739	FlangeDelayRoom	709
740	SloFlangeDlyRoom	709
741	FlangeDlyBigHall	709
742	Flange Theatre	710
743	FlangeVerb Clav	710
744	FlangeVerb Gtr	710
745	Flange Hall	710
746	Flange Booth	710
747	Flange->LaserDly	711
748	FlangeTap Synth	708
749	Lazertag Flange	711
750	Flange->Pitcher	712
751	Flange->Shaper	713
752	Shaper->Flange	713
753	Warped Echoes	715
754	L:Flange R:Delay	715
755	StereoFlamDelay	715
756	2Dlys Ch Fl Mono	716
757	LaserDelay->Rvb	717
758	Shaper->Reverb	718
759	MnPitcher+Chorus	720
760	MnPitcher+Flange	721
761	Pitcher+Chor+Dly	722
762	Pitcher+Flng+Dly	723
763	SubtleDistortion	724
764	Synth Distortion	727
765	Dist Cab EPiano	725
766	Distortion+EQ	726
767	Burnt Transistor	728
768	TubeAmp DlyChor	729
769	TubeAmp DlyChor2	729
770	TubeAmp DlyFlnge	730
771	TubeAmp Flange	730
772	PolyAmp Chorus	731
773	PolyAmp DlyFlnge	732
774	VibrChor Rotors	733
775	SlightDistRotors	734
776	Rotostort	734
777	VibrChor Rotors2	733
778	Full VbCh Rotors	737
779	KB3 FXBus	735
780	KB3 AuxFX	736
900	Basic Env Filter	900
901	Phunk Env Filter	900
902	Synth Env Filter	900
903	Bass Env Filter	900
904	EPno Env Filter	900
905	Trig Env Filter	901
906	LFO Sweep Filter	902
907	DoubleRiseFilter	902
908	Circle Bandsweep	902
909	Resonant Filter	903
910	Dual Res Filter	904
911	EQ Morpher	905
912	Mono EQ Morpher	906
913	Ring Modulator	907
914	PitcherA	908
915	PitcherB	908
916	SuperShaper	909
917	SubtleDrumShape	910
918	3 Band Shaper	910
919	LaserVerb	913
920	Laserwaves	913
921	Crystallizer	913
922	Spry Young Boy	912
923	Cheap LaserVerb	912
924	Drum Neurezonate	911
925	LazerfazerEchoes	911
950	HKCompressor 3:1	950
951	DrumKompress 5:1	950
952	SK FB Comprs 6:1	951
953	SKCompressor 9:1	951
954	SKCompressr 12:1	951
955	Compress w/SC EQ	953
956	Compress/Expand	954
957	Comprs/Expnd +EQ	955
958	Reverb>Compress	719
959	Reverb>Compress2	719
960	Drum Comprs>Rvb	719
961	Expander	952
962	3Band Compressor	956
963	Simple Gate	957
964	Gate w/ SC EQ	958
965	Graphic EQ	968
966	5 Band EQ	970
967	ContourGraphicEQ	969
968	Dance GraphicEQ	969
969	OldPianoEnhancer	959
970	3 Band Enhancer	960
971	3 Band Enhancer2	960
972	Extreem Enhancer	960
973	Tremolo	962
974	Dual Panner	964
975	SRS	965
976	Widespread	966
977	Mono->Stereo	967
998	Stereo Analyze	999
999	FX Mod Diag	998

KDFX Studios

ID	Name	Bus1 FX Preset	Bus2 FX Preset	Bus3 FX Preset	Bus4 FX Preset	Aux Bus FX Preset
1	RoomChorDly Hall	16	156	714	0	78
2	RmChorChRv Hall	17	154	722	0	69
3	RoomChorCDR Hall	16	156	714	0	76
4	RoomChor Hall	23	157	0	0	78
5	RoomChrCh4T Hall	22	156	706	0	72
6	RoomFlngCDR Hall	42	170	711	0	75
7	RoomFlgEcho Hall	21	176	131	0	85
8	RmFlngStImg Garg	19	172	976	0	45
9	RmFlgChDly Room	20	172	151	0	24
10	ChmbFlgGtRv Hall	42	170	112	0	75
11	RoomFlngCDR Hall	16	172	718	0	87
12	RoomFlngLsr Echo	22	172	925	0	119
13	RmFlgFXFlng Flng	23	174	173	0	171
14	SpaceFlng Hall	58	170	0	0	30
15	ChmbFlngCDR Verb	42	170	711	0	83
16	RoomPhsrCDR Hall	16	190	712	0	76
17	RmPhsrQuFlg Hall	19	190	737	0	76
18	RoomPhsr Space	25	191	0	0	114
19	RmEQmphEcho Comp	17	912	131	0	954
20	RmEQmphEcho Hall	17	912	131	0	65
21	RmEQmph4Tp Space	17	912	133	0	5
22	RmEQmph4Tap Hall	17	912	133	0	65
23	RmSweepEcho Hall	15	906	130	0	69
24	RoomResEcho Hall	3	909	131	0	71
25	RmRotoFl4T CmpRv	15	777	734	0	959
26	RoomSrsCDR Hall	16	975	712	0	75
27	RoomSRSRoom Room	17	975	15	0	29
28	RoomSRSChDl Hall	22	975	700	0	78
29	RoomSrsCDR CDR	16	975	712	0	711
30	RmStImgChDl Hall	22	976	700	0	73
31	RoomSRSRoom Chmb	17	975	15	0	47
32	RoomSRSRoom Hall	17	975	15	0	78
33	ChmbCompCDR Hall	42	953	711	0	75
34	RoomCmpChor Hall	15	951	152	0	78
35	RoomComp Hall	27	951	0	0	79
36	RoomComp Hall	7	953	0	0	67
37	BthComp SRS Hall	2	952	0	975	63
38	RoomCmpCh4T Hall	23	951	706	0	78
39	RmDsRotFl4t RvCm	15	776	734	0	959
40	RoomRmHall Hall	22	17	55	0	100
41	Room Room SRS2	22	0	44	0	975
42	RoomRmHall Hall	22	17	55	0	78
43	Room Room Hall	22	0	44	0	75
44	Room Hall Hall	23	0	61	0	78
45	Room Room Hall2	22	0	23	0	79
46	Room Room Hall2	22	44	0	0	85
47	Room Room Hall2	22	0	44	0	85
48	Room Hall Hall2	22	0	62	0	85
49	Sndbrd Room Hall	6	0	15	0	68
50	Sndbrd Rm Hall2	6	0	15	0	73
51	Room Room Hall3	22	0	15	0	68
52	auxChrMDly Room	0	158	753	0	30
53	auxFlngChRv Room	0	170	723	0	28
54	auxShp4MDly Hall	0	917	756	0	63
55	auxDistLasr Room	0	763	920	0	29
56	auxEnhSp4T Class	0	970	136	0	19
57	auxDistLasr Acid	0	767	924	0	118
58	EnhcManPhs Room	970	192	0	0	27
59	EnhrFlg8Tap Room	969	170	135	0	15
60	EnhcCmpFlng Room	969	950	177	0	24

ID	Name	Bus1 FX Preset	Bus2 FX Preset	Bus3 FX Preset	Bus4 FX Preset	Aux Bus FX Preset
61	CompEQmphCh Room	952	912	153	0	4
62	BthQFlg4Tap Hall	2	737	133	0	76
63	ChmbTremCDR Room	42	973	715	0	29
64	ChmbCmpFlRv Hall	41	952	744	0	69
65	ChamDstEcho Room	41	764	131	0	28
66	ChamFlg4Tap Hall	41	173	136	0	75
67	ChmbEnv4Tap GtRv	42	903	134	0	112
68	CmbrShapLsr Hall	42	916	922	0	69
69	auxPtchDst+ Chmb	0	914	772	0	48
70	auxChorFlRv Cmbr	0	150	742	0	42
71	auxChorFlRv Cmb2	0	155	742	0	42
72	auxChorFlRv Cmb3	0	150	745	0	42
73	auxChorFlRv Cmb4	0	150	742	0	18
74	HallFlgChDl Room	56	177	700	0	29
75	HallPtchLsr Hall	57	915	922	0	75
76	HallGateFl4T Bth	55	963	748	0	1
77	HallChorFDR Room	55	707	739	0	29
78	HallPtchPtFl Lsr	57	915	760	0	919
79	HallFlng8Tp Room	56	176	135	0	29
80	HallChrEcho Room	55	158	132	0	31
81	HallChorCDR Hall	55	152	715	0	55
82	HallRsFltChDl Rm	46	909	700	0	18
83	Hall ChDly Hall	56	0	704	0	30
84	HallFlgChDl Hall	56	177	700	0	65
85	Hall Room SRS	75	0	17	0	975
86	Hall Room Room	78	0	15	0	22
87	Hall CmpRvb	67	0	0	0	958
88	Hall Flng Hall	63	177	0	0	86
89	HallRoomChr Hall	46	15	151	0	82
90	auxPhsrFDR Hall	0	193	741	0	75
91	auxChrDist+ Hall	0	150	768	0	75
92	auxFlgDist+ Hall	0	170	769	0	75
93	auxChrDst+ Hall	0	150	768	0	76
94	auxChorMDly Hall	0	159	755	0	76
95	auxChorSp6T Hall	0	152	138	0	75
96	auxChorChDl Hall	0	153	702	0	64
97	auxPhasStIm Hall	0	195	976	0	95
98	auxFlngCDR Hall	0	172	713	0	65
99	auxPhsrFldblHall	0	193	175	0	75
100	auxSRSRoom Hall	0	975	25	0	78
101	auxFlLsr SwHall	0	170	922	0	72
102	auxEnh4Tap Hall	0	972	133	0	79
103	EnhcChorCDR Hall	969	152	716	0	56
104	EnhChorChDl Hall	970	156	703	0	61
105	EnhcChor Plate	971	152	0	0	98
106	CompFlgChor Hall	952	173	153	0	63
107	ChorChorFlg Hall	159	150	170	0	55
108	ChapelSRS Hall	60	975	0	0	79
109	ChapelSRS Hall2	60	975	0	0	85
110	Chapel Room Hall	60	0	23	0	78
111	PltEnvFl4T Room	43	903	735	0	25
112	PlatEnvFl4T Filt	43	903	735	0	907
113	PltEnvFl4T Plate	43	902	735	0	103
114	PltTEnvFlg Plate	43	905	170	0	31
115	PlateRngMd Hall	102	913	0	0	95
116	auxDist+Echo Plt	0	772	130	0	31
117	auxEnvSp4T Plate	0	904	136	0	31
118	auxShap4MD Plate	0	918	756	0	31
119	auxChorDist+ Plt	0	156	768	0	31
120	auxShFlgChDl Plt	0	752	710	0	103

ID	Name	Bus1 FX Preset	Bus2 FX Preset	Bus3 FX Preset	Bus4 FX Preset	Aux Bus FX Preset
121	auxMPFlgLasr Plt	0	760	923	0	103
122	auxShap4MD Plate	0	917	756	0	31
123	FlgEnv4Tap Plate	173	904	133	0	31
124	EnhrFlgCDR Plate	969	170	712	0	96
125	auxRingPFD Plate	0	913	762	0	97
126	GtRvShapMDl Room	112	916	754	0	29
127	GtdEnhcStIm Room	112	969	976	0	17
128	Gtd2ChrEcho 2Vrb	112	151	130	0	110
129	GtdEnhcStIm Hall	112	969	976	0	72
130	auxEnvSp4T GtVrb	0	904	136	0	112
131	GtRbSwpFlt Lasr	112	908	0	0	924
132	GtRbSwpFlt FlDly	112	907	0	0	733
133	ChRvStIEcho Hall	724	976	130	0	75
134	ChorChorCDR Spac	151	152	715	0	58
135	ChDlDstEQ Hall	701	767	0	0	83
136	auxDPanCDR ChPlt	0	974	713	0	725
137	AuxChorFlng CDR	0	157	173	0	712
138	auxEnhcSp4T CDR	0	970	136	0	711
139	auxPtchDst+ ChRv	0	914	772	0	721
140	EnhcChorChDl PCD	970	156	703	0	761
141	auxPoly FDR	0	764	0	0	738
142	EnhcChorChDl FDR	970	156	703	0	740
143	EnhcChrChDl FDR2	970	156	705	0	740
144	auxRotoSp4T FlRv	0	777	136	0	743
145	auxRotaryFDR Plt	0	774	739	0	97
146	RotoOrgFX Hall	778	0	0	0	59
147	CmpRvbFlDl Hall	960	0	732	0	86
148	auxEnhSp4T CmpRv	0	971	136	0	958
149	auxPtchRoom RvCm	0	914	17	0	958
150	PhsrChorCDR Phsr	194	151	717	0	194
151	ChDlSp4TFlDl Phs	151	137	732	0	192
152	auxFlgDst+ ChLsD	0	170	769	0	709
153	auxFlgDst+ ChLs2	0	170	771	0	709
154	RoomRoomSRS CmRv	4	15	0	975	960
155	RoomRoom Room	5	18	0	0	27
156	GtRvPlate Hall	113	96	0	0	82
157	RoomRoom SRS	17	26	0	0	975
158	EnhcSp4T Hall	970	136	0	0	61
159	Room RoomChr SRS	17	0	15	157	975
160	KB3 V/C ->Rotary	779	0	0	0	780
161	EQStImg 5BndEQ	199	965	976	199	966
162	aux5BeqStIm Hall	199	966	976	199	78
198	Digitech Studio	0	0	0	0	0
199	Default Studio	0	0	0	0	0

KDFX Algorithm Specifications

Algorithms 1 and 2: MiniVerbs

1 MiniVerb
2 Dual MiniVerb

Versatile, small stereo and dual mono reverbs

PAUs: 1 for MiniVerb
2 for Dual MiniVerb

MiniVerb is a versatile stereo reverb is found in many combination algorithms, but is equally useful on its own because of its small size. The main control for this effect is the Room Type parameter. Room Type changes the structure of the algorithm to simulate many carefully crafted room types and sizes. Spaces characterized as booths, small rooms, chambers, halls and large spaces can be selected.

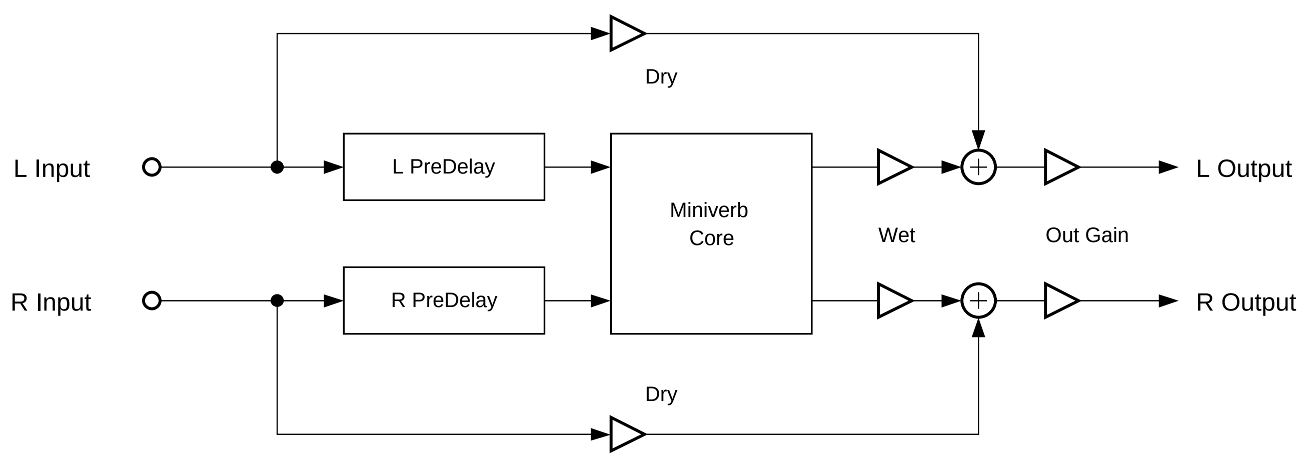


Figure 10-1 Simplified Block Diagram of MiniVerb

Each Room Type incorporates different diffusion, room size and reverb density settings. The Room Types were designed to sound best when Diff Scale, Size Scale and Density are set to the default values of **1.00x**. If you want a reverb to sound perfect immediately, set the Diff Scale, Size Scale and Density parameters to **1.00x**, pick a Room Type and you'll be on the way to a great sounding reverb. But if you want to experiment with new reverb flavors, changing the scaling parameters away from **1.00x** can cause a subtle (or drastic!) coloring of the carefully crafted Room Types.

Diffusion characterizes how the reverb spreads the early reflection out in time. At very low settings of Diff Scale, the early reflections start to sound quite discrete, and at higher settings the early reflections are seamless. Density controls how tightly the early reflections are packed in time. Low Density settings have the early reflections grouped close together, and higher values spread the reflections for a smoother reverb.

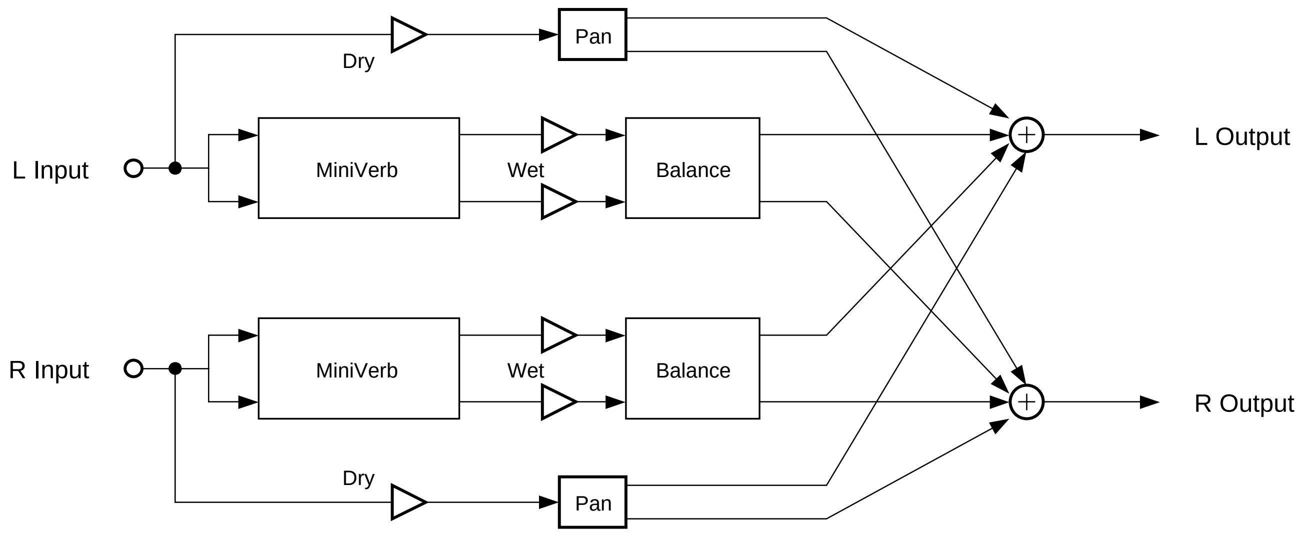


Figure 10-2 Simplified Block Diagram of Dual MiniVerb

Dual MiniVerb has a full MiniVerb, including Wet/Dry, Pre Delay and Out Gain controls, dedicated to both the left and right channels. In Figure 10-2, the two blocks labeled MiniVerb contain a complete copy of the contents of Figure 10-1. Dual MiniVerb gives you indepenent reverbs on both channels which has obvious benefits for mono material. With stereo material, any panning or image placement can be maintained, even in the reverb tails! This is pretty unusual behaviour for a reverb, since even real halls will rapidly delocalize acoustic images in the reverberance. Since maintaining image placement in the reverberation is so unusual, you will have to carefully consider whether it is appropriate for your particular situation. To use Dual MiniVerb to maintain stereo signals in this manner, set the reverb parameters for both channels to the same values. The Dry Pan and Wet Bal parameters should be fully left (**-100%**) for the left MiniVerb and fully right (**100%**) for the right MiniVerb.

MiniVerb Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Rvrb Time	0.5 to 30.0 s, Inf	HF Damping	16 to 25088 Hz
L Pre Dly	0 to 620 ms	R Pre Dly	0 to 620 ms

Page 2

Room Type	Hall1	Diff Scale	0.00 to 2.00x
		Size Scale	0.00 to 4.00x
		Density	0.00 to 4.00x

Dual MiniVerb Parameters

Page 1

L Wet/Dry	0 to 100%wet	R Wet/Dry	0 to 100%wet
L Out Gain	Off, -79.0 to 24.0 dB	R Out Gain	Off, -79.0 to 24.0 dB
L Wet Bal	-100 to 100%	R Wet Bal	-100 to 100%
L Dry Pan	-100 to 100%	R Dry Pan	-100 to 100%

Page 2

L RoomType	Hall1		
L RvrbTime	0.5 to 30.0 s, Inf		
L Diff Scl	0.00 to 2.00x	L Density	0.00 to 4.00x
L Size Scl	0.00 to 4.00x	L HF Damp	16 to 25088 Hz
L PreDlyL	0 to 620 ms	L PreDlyR	0 to 620 ms

Page 3

R RoomType	Hall1		
R RvrbTime	0.5 to 30.0 s, Inf		
R Diff Scl	0.00 to 2.00x	R Density	0.00 to 4.00x
R Size Scl	0.00 to 4.00x	R HF Damp	16 to 25088 Hz
R PreDlyL	0 to 620 ms	R PreDlyR	0 to 620 ms

Wet/Dry A simple mix of the reverb sound with the dry sound.

Out Gain The overall gain or amplitude at the output of the effect.

Rvrb Time The reverb time displayed is accurate for normal settings of the other parameters (HF Damping = 25088kHz, and Diff Scale, Room Scale and Density = 1.00x). Changing Rvrb Time to Inf creates an infinitely sustaining reverb.

HF Damping Reduces high frequency components of the reverb above the displayed cutoff frequency. Removing higher reverb frequencies can often make rooms sound more natural.

L/R Pre Dly The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer pre-delays can help make larger spaces sound more realistic. Longer times can also help improve the clarity of a mix by separating the reverb signal from the dry signal, so the dry signal is not obscured. Likewise, the wet signal will be more audible if delayed, and thus you can get by with a dryer mix while maintaining the same subjective wet/dry level.

Room Type Changes the configuration of the reverb algorithm to simulate a wide array of carefully designed room types and sizes. This parameter effectively allows you to have several different reverb algorithms only a parameter change away. Smaller Room Types will sound best with shorter Rvrb Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you don’t want to modulate it.)

Diff Scale A multiplier which affects the diffusion of the reverb. At 1.00x, the diffusion will be the normal, carefully adjusted amount for the current Room Type. Altering this parameter will change the diffusion from the preset amount.

Size Scale A multiplier which changes the size of the current room. At 1.00x, the room will be the normal, carefully tweaked size of the current Room Type. Altering this parameter will change the size of the room, and thus will cause a subtle coloration of the reverb (since the room's dimensions are changing).

Density A multiplier which affects the density of the reverb. At 1.00x, the room density will be the normal, carefully set amount for the current Room Type. Altering this parameter will change the density of the reverb, which may color the room slightly.

Wet Bal In Dual MiniVerb, two mono signals (left and right) are fed into two separate stereo reverbs. If you center the wet balance (0%), the left and right outputs of the reverb will be sent to the final output in equal amounts. This will add a sense of spaciousness

3 Gated MiniVerb

A reverb and compressor in series.

PAUs: 2

This algorithm is a small reverb followed by a gate. The main control for the reverb is the Room Type parameter. The main control for the reverb is the Room Type parameter. Room Type changes the structure of the algorithm to simulate many carefully crafted room types and sizes. Spaces characterized as booths, small rooms, chambers, halls and large spaces can be selected.

Each Room Type incorporates different diffusion, room size and reverb density settings. The Room Types were designed to sound best when Diff Scale, Size Scale and Density are set to the default values of 1.00x. If you want a reverb to sound perfect immediately, set the Diff Scale, Size Scale and Density parameters to 1.00x, pick a Room Type and you'll be on the way to a great sounding reverb. But if you want experiment with new reverb flavors, changing the scaling parameters away from 1.00x can cause a subtle (or drastic!) coloring of the carefully crafted Room Types.

Diffusion characterizes how the reverb spreads the early reflection out in time. At very low settings of Diff Scale, the early reflections start to sound quite discrete, and at higher settings the early reflections are seamless. Density controls how tightly the early reflections are packed in time. Low Density settings have the early reflections grouped close together, and higher values spread the reflections for a smoother reverb.

The gate turns the output of the reverb on and off based on the amplitude of the input signal.

A gate behaves like an on off switch for a signal. One or both input channels is used to control whether the switch is on (gate is open) or off (gate is closed). The on/off control is called "side chain" processing. You select which of the two input channels or both is used for side chain processing. When you select both channels, the sum of the left and right input amplitudes is used. The gate is opened when the side chain amplitude rises above a level that you specify with the Threshold parameter.

The gate will stay open for as long as the side chain signal is above the threshold. When the signal drops below the threshold, the gate will remain open for the time set with the Gate Time parameter. At the end of the Gate Time, the gate closes. When the signal rises above threshold, it opens again. What is happening is that the gate timer is being constantly retriggered while the signal is above threshold.

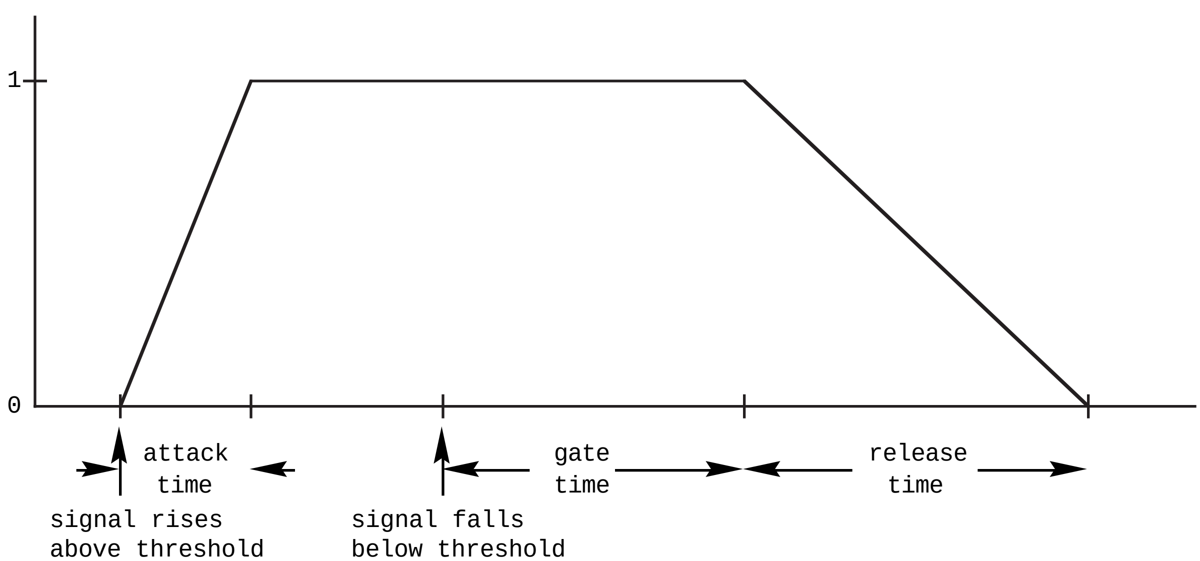


Figure 10-3 Gate Behavior

If Gate Duck is turned on, then the behaviour of the gate is reversed. The gate is open while the side chain signal is below threshold, and it closes when the signal rises above thresold.

If the gate opened and closed instantaneously, you would hear a large digital click, like a big knife switch was being thrown. Obviously that's not a good idea, so Gate Atk (attack) and Gate Rel (release) parameters are use to set the times for the gate to open and close. More precisely, depending on whether Gate Duck is off or on, Gate Atk sets how fast the gate opens or closes when the side chain signal rises above the threshold. The Gate Rel sets how fast the gate closes or opens after the gate timer has elapsed.

The Signal Dly parameter delays the signal being gated, but does not delay the side chain signal. By delaying the main signal relative to the side chain signal, you can open the gate just before the main signal rises above threshold. It's a little like being able to pick up the telephone before it rings!

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Rvrb Time	0.5 to 30.0s, Inf	HF Damping	16 to 25088 Hz
L Pre Dly	0 to 620ms	R Pre Dly	0 to 620 ms

Page 2

Room Type	Hall1	Diff Scale	0.00 to 2.00x
		Size Scale	0.00 to 4.00x
		Density	0.00 to 4.00x

Page 3

Gate Thres	-79.0 to 0.0 dB	Gate Time	0 to 3000 ms
Gate Duck	In or Out	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		Reduction	-dB 60 40 * 16 * 8 4 0

Wet/Dry A simple mix of the reverb sound with the dry sound. When set fully dry (0%), the gate is still active.

Out Gain An overall level control of the effect's output (applied after the Wet/Dry mix).

Rvrb Time The reverb time displayed is accurate for normal settings of the other parameters (HF Damping = 25088kHz, and Diff Scale, Room Scale and Density = 1.00x). Changing Rvrb Time to Inf creates an infinitely sustaining reverb.

HF Damping Reduces high frequency components of the reverb above the displayed cutoff frequency. Removing higher reverb frequencies can often make rooms sound more natural.

L/R Pre Dly The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer pre-delays can help make larger spaces sound more realistic. Longer times can also help improve the clarity of a mix by separating the reverb signal from the dry signal, so the dry signal is not obscured. Likewise, the wet signal will be more audible if delayed, and thus you can get by with a dryer mix while maintaining the same subjective wet/dry level.

Room Type The configuration of the reverb algorithm to simulate a wide array of carefully designed room types and sizes. This parameter effectively allows you to have several different reverb algorithms only a parameter change away. Smaller Room Types will sound best with shorter Rvrb Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you may not modulate it.)

Diff Scale A multiplier which affects the diffusion of the reverb. At 1.00x, the diffusion will be the normal, carefully adjusted amount for the current Room Type. Altering this parameter will change the diffusion from the preset amount.

Size Scale A multiplier which changes the size of the current room. At 1.00x, the room will be the normal, carefully tweaked size of the current Room Type. Altering this parameter will change the size of the room, and thus will cause a subtle coloration of the reverb (since the room's dimensions are changing).

Density A multiplier which affects the density of the reverb. At 1.00x, the room density will be the normal, carefully set amount for the current Room Type. Altering this parameter will change the density of the reverb, which may color the room slightly.

Gate Thres The input signal level in dB required to open the gate (or close the gate if Gate Duck is on).

Gate Duck When set to "Off", the gate opens when the signal rises above threshold and closes when the gate time expires. When set to "On", the gate closes when the signal rises above threshold and opens when the gate time expires.

Gate Time The time in seconds that the gate will stay fully on after the signal envelope rises above threshold. The gate timer is started or restarted whenever the signal envelope rises above threshold. If Retrigger is On, the gate timer is continually reset while the side chain signal is above the threshold.

Gate Atk The attack time for the gate to ramp from closed to open (reverse if Gate Duck is on) after the signal rises above threshold.

Gate Rel The release time for the gate to ramp from open to closed (reverse if Gate Duck is on) after the gate timer has elapsed.

Signal Dly The delay in milliseconds (ms) of the reverb signal relative to the side chain signal. By delaying the reverb signal, the gate can be opened before the reverb signal rises above the gating threshold.

Algorithms 4–11: Classic / TQ / Diffuse / Omni Reverbs

4 Classic Place
5 Classic Verb
6 TQ Place
7 TQ Verb
8 Diffuse Place
9 Diffuse Verb
10 OmniPlace
11 OmniVerb

Parameters

Absorption This controls the amount of reflective material that is in the space being emulated, much like an acoustical absorption coefficient. The lower the setting, the longer it will take for the sound to die away. A setting of 0% will cause an infinite decay time.

Rvrb Time Adjusts the basic decay time of the late portion of the reverb.

LateRvbTim Adjusts the basic decay time of the late portion of the reverb after diffusion.

HF Damping This controls the amount of high frequency energy that is absorbed as the reverb decays. The values set the cutoff frequency of the 1 pole (6dB/oct) lopass filter within the reverb feedback loop.

L Pre Dly, R Pre Dly These control the amount that each channel of the reverb is delayed relative to the dry signal. Setting different lengths for both channels can de-correlate the center portion of the reverb image and make it seem wider. This only affects the late reverb in algorithms that have early reflections.

Lopass Controls the cutoff frequency of a 1 pole (6dB/oct) lopass filter at the output of the reverb. This only affects the late reverb in algorithms that have early reflections.

EarRef Lvl Adjusts the mix level of the early reflection portion of algorithms offering early reflections.

Late Lvl Adjusts the mix level of the late reverb portion of algorithms offering early reflections.

Room Type This parameter selects the basic type of reverb being emulated, and should be your starting point when creating your own reverb presets. Due to the inherent complexity of reverb algorithms and the sheer number of variables responsible for their character, the Room Type parameter provides condensed preset collections of these variables. Each Room Type preset has been painstakingly selected by Kurzweil engineers to provide the best sounding collection of mutually complementary variables modelling an assortment of reverb families. When a room type is selected, an entire incorporated set of delay lengths and diffusion settings are established within the algorithm. By using the Size Scale, DiffAmtScl, DiffLenScl, and Inj Spread parameters, you may scale individual elements away from their preset value. When set to 1.00x, each of these elements are accurately representing their preset values determined by the current Room Type.

Room Types with similar names in different reverb algorithms do not sound the same. For example, Hall1 in Diffuse Verb does not sound the same as Hall1 in TQ Verb.

Size Scale This parameter scales the inherent size of the reverb chosen by Room Type. For a true representation of the selected Room Type size, set this to 1.00x. Scaling the size below this will create smaller spaces, while larger scale factors will create large spaces. See Room Type for more detailed information.

InfinDecay Found in "Verb" algorithms. When turned "On", the reverb tail will decay indefinitely. When turned "Off", the decay time is determined by the "Rvrb Time" or "LateRvbTim" parameters.

LF Split Used in conjunction with LF Time. This controls the upper frequency limit of the low frequency decay time multiplier. Energy below this frequency will decay faster or slower depending on the LF Time parameter.

LF Time Used in conjunction with LF Split. This modifies the decay time of the energy below the LF Split frequency. A setting of 1.00x will make low frequency energy decay at the rate determined by the decay time. Higher values will cause low frequency energy to decay slower, and lower values will cause it to decay more quickly.

TrebShlf F Adjusts the frequency of a high shelving filter at the output of the late reverb.

TrebShlf G Adjusts the gain of a high shelving filter at the output of the late reverb.

BassShlf F Adjusts the frequency of a low shelving filter at the output of the late reverb.

BassShlf G Adjusts the gain of a low shelving filter at the output of the late reverb.

DiffAmtScl Adjusts the amount of diffusion at the onset of the reverb. For a true representation of the selected Room Type diffusion amount, set this to 1.00x.

DiffLenScl Adjusts the length of the diffusion at the onset of the reverb. For a true representation of the selected Room Type diffusion length, set this to 1.00x.

DiffExtent Adjust the onset diffusion duration. Higher values create longer diffuse bursts at the onset of the reverb.

Diff Cross Adjusts the onset diffusion cross-coupling character. Although subtle, this parameter bleeds left and right channels into each other during onset diffusion, and also in the body of the reverb. 0% setting will disable this. Increasing this value in either the positive or negative direction will increase its affect.

Expanse Amount of late reverb energy biased toward the edges of the stereo image. A setting of 0% will bias energy towards the center. Moving away from 0% will bias energy towards the sides. Positive and negative values will have a different character.

LFO Rate Adjusts the rate at which certain reverb delay lines move. See LFO Depth for more information.

LFO Depth Adjusts the detuning depth in cents caused by a moving reverb delay line. Moving delay lines can imitate voluminous flowing air currents and reduce unwanted artifacts like ringing and flutter when used properly. Depth settings under 1.5ct with LFO Rate settings under 1.00Hz are recommended for modeling real spaces. High depth settings can create chorusing qualities, which won't be unsuitable for real acoustic spaces, but can nonetheless create interesting effects. Instruments that have little if no inherent pitch fluctuation (like piano) are much more sensitive to this LFO than instruments that normally have a lot of vibrato (like voice) or non-pitched instruments (like snare drum).

Inj Build Used in conjunction with Inj Spread, this adjusts the envelope of the onset of the reverb. Specifically, it tapers the amplitudes of a series of delayed signals injected into the body of the reverb. Values above 0% will produce a faster build, while values below 0% will cause the build to be more gradual.

Inj Spread Used in conjunction with Inj Build, this scales the length of the series of delays injected into the body of the reverb. For a true representation of the selected Room Type injector spread, set this to 1.00x.

Inj LP This adjusts the cutoff frequency of a 1 pole (6dB/oct) lopass filter applied to the signal being injected into the body of the reverb.

Inj Skew Adjusts the amount of delay applied to either the left or right channel of the reverb injector. Positive values delay the right channel while negative values delay the left channel.

E DiffAmt Adjusts the amount of diffusion applied to the early reflection network.

E DfLenScl Adjusts the length of diffusion applied to the early reflection network. This is influenced by E PreDlyL and E PreDlyR.

E Dly Scl Scales the delay lengths inherent in the early reflection network.

E Build Adjusts the envelope of the onset of the early reflections. Values above 0% will create a faster attack while values below 0% will create a slower attack.

E Fdbk Amt Adjusts the amount of the output of an early reflection portion that is fed back into the input of the opposite channel in front of the early pre-delays. Overall, it lengthens the decay rate of the early reflection network. Negative values polarity invert the feedback signal.

E HF Damp This adjusts the cutoff frequency of a 1 pole (6dB/oct) lopass filter applied to the early reflection feedback signal.

E PreDlyL, E PreDlyR Adjusts how much the early reflections are delayed relative to the dry signal. These are independent of the late reverb predelay times, but will influence E Dly Scl.

E Dly L, E Dly R Adjusts the left and right early reflection delays fed to the same output channels.

E Dly LX, E Dly RX Adjusts the left and right early reflection delays fed to the opposite output channels.

E DifDlyL, E DifDlyR Adjusts the diffusion delays of the diffusers on delay taps fed to the same output channels.

E DifDlyLX, E DifDlyRX Adjusts the diffusion delays of the diffusers on delay taps fed to the opposite output channels.

E X Blend Adjusts the balance between early reflection delay tap signals with diffusers fed to their same output channel, and those fed to opposite channels. 0% will only allow delay taps being fed to opposite output channels to be heard, while 100% allows only delay taps going to the same channels to be heard.

12 Panaural Room

Room reverberation algorithm

PAUs: 3

The Panaural Room reverberation is implemented using a special network arrangement of many delay lines that guarantees colorless sound. The reverberator is inherently stereo with each input injected into the "room" at multiple locations. The signals entering the reverberator first pass through a shelving bass equalizer with a range of +/-15dB. To shorten the decay time of high frequencies relative to mid frequencies, low pass filters controlled by HF Damping are distributed throughout the network. Room Size scales all the delay times of the network (but not the Pre Dly or Build Time), to change the simulated room dimension over a range of 1 to 16m. Decay Time varies the feedback gains to achieve decay times from 0.5 to 100 seconds. The Room Size and Decay Time controls are interlocked so that a chosen Decay Time will be maintained while Room Size is varied. A two input stereo mixer, controlled by Wet/Dry and Out Gain, feeds the output.

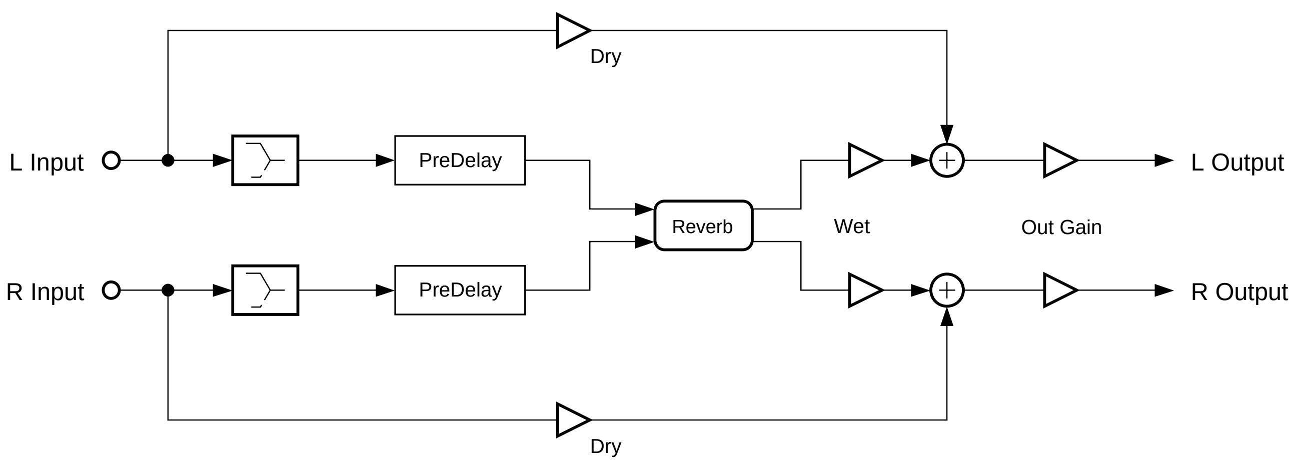


Figure 10-4 Simplified block diagram of Panaural Room.

The duration and spacing of the early reflections are influenced by Room Size and Build Time, while the number and relative loudness of the individual reflections are influenced by Build Env. When Build Env is near 0 or 100%, fewer reflections are created. The maximum number of important early reflections, 13, is achieved at a setting of 50%.

To get control over the growth of reverberation, the left and right inputs each are passed through an "injector" that can extend the source before it drives the reverberator. Only when Build Env is set to 0% is the reverberator driven in pure stereo by the pure dry signal. For settings of Build Env greater than 0%, the reverberator is fed multiple times. Build Env controls the injector so that the reverberation begins abruptly (0%), builds immediately to a sustained level (50%), or builds gradually to a maximum (100%). Build Time varies the injection length over a range of 0 to 500ms. At a Build Time of 0ms, there is no extension of the build time. In this case, the Build Env control adjusts the density of the reverberation, with maximum density at a setting of 50%. In addition to the two build controls, there is an overall Pre Dly control that can delay the entire reverberation process by up to 500ms.

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0
Room Size	1.0 to 16.0 m		
Pre Dly	0 to 500 ms	Decay Time	0.5 to 100.0 s
HF Damping	16 to 25088 Hz		

Page 2

Bass Gain	-15 to 15 dB	Build Time	0 to 500 ms
		Build Env	0 to 100%

Wet/Dry The amount of the stereo reverberator (wet) signal relative to the original input (dry) signal to be output. The dry signal is not affected by the Bass Gain control. The wet signal is affected by the Bass Gain control and by all the other reverberator controls. The balance between wet and dry signals is an extremely important factor in achieving a good mix. Emphasizing the wet signal gives the effect of more reverberation and of greater distance from the source.

Out Gain The overall output level for the reverberation effect, and controls the level for both the wet and dry signal paths.

Decay Time The reverberation decay time (mid-band "RT60"), the time required before the reverberation has died away to 60dB below its "running" level. Adjust decay time according to the tempo and articulation of the music and to taste.

HF Damping Adjusts low pass filters in the reverberator so that high frequencies die away more quickly than mid and low frequencies. This shapes the reverberation for a more natural, more acoustically accurate sound.

Bass Gain Shapes the overall reverberation signal's bass content, but does not modify the decay time. Reduce the bass for a less muddy sound, raise it slightly for a more natural acoustic effect.

Room Size Choosing an appropriate room size is very important in getting a good reverberation effect. For impulsive sources, such as percussion instruments or plucked strings, increase the size setting until discrete early reflections become audible, and then back it off slightly. For slower, softer music, use the largest size possible. At lower settings, Room Size leads to coloration, especially if the Decay Time is set too high.

Pre Dly Introducing predelay creates a gap of silence between that allows the dry signal to stand out with greater clarity and intelligibility against the reverberant background. This is especially helpful with vocal or classical music.

Build Time Similar to predelay, but more complex, larger values of Build Time slow down the building up of reverberation and can extend the build up process. Experiment with Build Time and Build Env and use them to optimize the early details of reverberation. A Build Time of 0ms and a Build Env of 50% is a good default setting that yields a fast arriving, maximally dense reverberation.

Build Env When Build Time has been set to greater than about 80ms, Build Env begins to have an audible influence on the early unfolding of the reverberation process. For lower density reverberation that starts cleanly and impulsively, use a setting of 0%. For the highest density reverberation, and for extension of the build up period, use a setting of 50%. For an almost reverse reverberation, set Build Env to 100%. You can think of Build Env as setting the position of a see-saw. The left end of the see-saw represents the driving of the reverberation at the earliest time, the pivot point as driving the reverberation at mid-point in the time sequence, and the right end as the last signal to drive the reverberator. At settings near 0%, the see-saw is tilted down on the right: the reverberation starts abruptly and the drive drops with time. Near 50%, the see-saw is level and the reverberation is repetitively fed during the entire build time. At settings near 100%, the see-saw is tilted down on the left, so that the reverberation is hit softly at first, and then at increasing level until the end of the build time.

13 Stereo Hall

A stereo hall reverberation algorithm.

PAUs: 3

The Stereo Hall reverberation is implemented using a special arrangement of all pass networks and delay lines which reduces coloration and increases density. The reverberator is inherently stereo with each input injected into the "room" at multiple locations. To shorten the decay time of low and high frequencies relative to mid frequencies, bass equalizers and low pass filters, controlled by Bass Gain and by HF Damping, are placed within the network. Room Size scales all the delay times of the network (but not the Pre Dly or Build Time), to change the simulated room dimension over a range of 10 to 75m. Decay Time varies the feedback gains to achieve decay times from 0.5 to 100 seconds. The Room Size and Decay Time controls are interlocked so that a chosen Decay Time will be maintained while Room Size is varied. At smaller sizes, the reverb becomes quite colored and is useful only for special effects. A two input stereo mixer, controlled by Wet/Dry and Out Gain, feeds the output. The Lowpass control acts only on the wet signal and can be used to smooth out the reverb high end without modifying the reverb decay time at high frequencies.

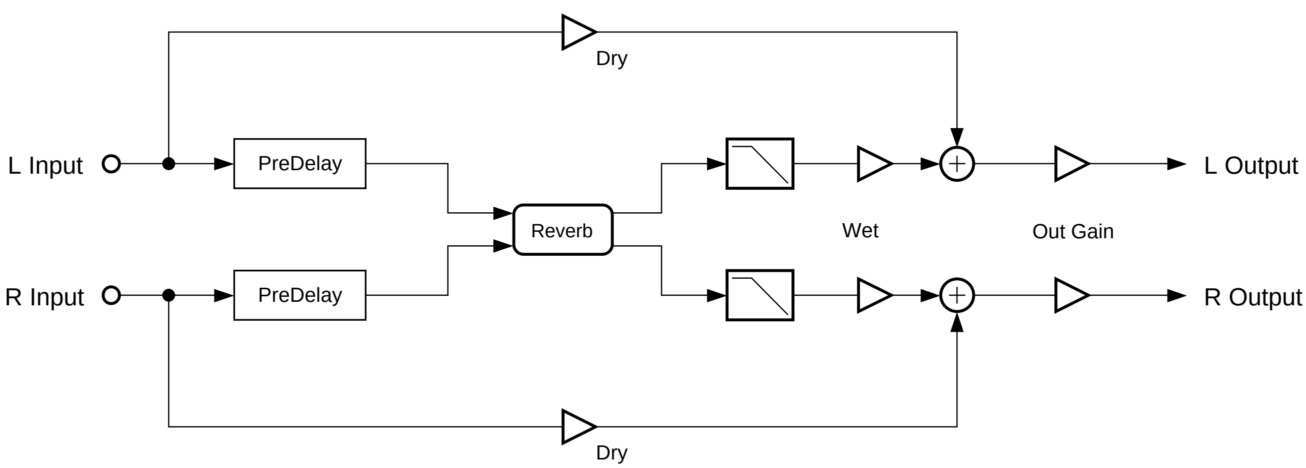


Figure 10-5 Simplified block diagram of Stereo Hall.

Within the reverberator, certain delays can be put into a time varying motion to break up patterns and to increase density in the reverb tail. Using the LFO Rate and Depth controls carefully with longer decay times can be beneficial. But beware of the pitch shifting artifacts which can accompany randomization when it is used in greater amounts. Also within the reverberator, the Diffusion control can reduce the diffusion provided by some all pass networks. While the reverb will eventually reach full diffusion regardless of the Diffusion setting, the early reverb diffusion can be reduced, which sometimes is useful to help keep the dry signal "in the clear".

The reverberator structure is stereo and requires that the dry source be applied to both left and right inputs. If the source is mono, it should still be applied (pan centered) to both left and right inputs. Failure to drive both inputs will result in offset initial reverb images and later ping-ponging of the reverberation. Driving only one input will also increase the time required to build up reverb density.

To gain control over the growth of reverberation, the left and right inputs each are passed through an "injector" that can extend the source before it drives the reverberator. Only when Build Env is set to 0% is the reverberator driven in pure stereo by the pure dry signal. For settings of Build Env greater than 0%, the reverberator is fed multiple times. Build Env controls the injector so that the reverberation begins abruptly (0%), builds immediately to a sustained level (50%), or builds gradually to a maximum (100%). Build Time varies the injection length over a range of 0 to 500ms. At a Build Time of 0ms, there is no extension of the build time. In this case, the Build Env control adjusts the density of the reverberation, with maximum density at a setting of 50%. In addition to the two build controls, there is an overall Pre Dly control that can delay the entire reverberation process by up to 500ms.

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Room Size	2.0 to 15.0 m	Diffusion	0 to 100%
Pre Dly	0 to 500 ms	Decay Time	0.5 to 100.0 ms
HF Damping	16 to 25088 Hz		

Page 2

Bass Gain	-15 to 0 dB	Build Time	0 to 500 ms
Lowpass	16 to 25088 Hz	Build Env	0 to 100%
LFO Rate	0.00 to 5.10 Hz		
LFO Depth	0.00 to 10.20 ct		

Wet/Dry The amount of the stereo reverberator (wet) signal relative to the original input (dry) signal to be output. The dry signal is not affected by the HF Roll control. The wet signal is affected by the HF Roll control and by all the other reverberator controls. The balance between wet and dry signals is an extremely important factor in achieving a good mix. Emphasizing the wet signal gives the effect of more reverberation and of greater distance from the source.

Out Gain The overall output level for the reverberation effect, and controls the level for both the wet and dry signal paths.

Decay Time The reverberation decay time (mid-band "RT60"), the time required before the reverberation has died away to 60dB below its "running" level. Adjust decay time according to the tempo and articulation of the music and to taste.

HF Damping Adjusts low pass filters in the reverberator so that high frequencies die away more quickly than mid and low frequencies. This shapes the reverberation for a more natural, more acoustically accurate sound.

Bass Gain Adjusts bass equalizers in the reverberator so that low frequencies die away more quickly than mid and high frequencies. This can be used to make the reverberation less muddy.

Lowpass Used to shape the overall reverberation signal's treble content, but does not modify the decay time. Reduce the treble for a softer, more acoustic sound.

Room Size Choosing an appropriate room size is very important in getting a good reverberation effect. For impulsive sources, such as percussion instruments or plucked strings, increase the size setting until discrete early reflections become audible, and then back it off slightly. For slower, softer music, use the largest size possible. At lower settings, RoomSize leads to coloration, especially if the DecayTime is set too high.

Pre Dly Introducing predelay creates a gap of silence between that allows the dry signal to stand out with greater clarity and intelligibility against the reverberant background. This is especially helpful with vocal or classical music.

Build Time Similar to predelay, but more complex, larger values of BuildTime slow down the building up of reverberation and can extend the build up process. Experiment with BuildTime and BuildEnv and use them to optimize the early details of reverberation. A BuildTime of 0ms and a BuildEnv of 0% is a good default setting that yields fast arriving, natural reverberation.

Build Env When BuildTime has been set to greater than about 80ms, BuildEnv begins to have an audible influence on the early unfolding of the reverberation process. For lower density reverberation that starts cleanly and impulsively, use a setting of 0%. For the highest density reverberation, and for extension of the build up period, use a setting of 50%. For an almost reverse reverberation, set BuildEnv to 100%. You can think of BuildEnv as setting the position of a see-saw. The left end of the see-saw represents the driving of the reverberation at the earliest time, the pivot point as driving the reverberation at mid-point in the time sequence, and the right end as the last signal to drive the reverberator. At settings near 0%, the see-saw is tilted down on the right: the reverberation starts abruptly and the drive drops with time. Near 50%, the see-saw is level and the reverberation is repetitively fed during the entire build time. At settings near 100%, the see-saw is tilted down on the left, so that the reverberation is hit softly at first, and then at increasing level until the end of the build time.

LFO Rate and Depth Within the reverberator, the certain delay values can be put into a time varying motion to break up patterns and to increase density in the reverb tail. Using the LFO Rate and Depth controls carefully with longer decay times can be beneficial. But beware of the pitch shifting artifacts which can accompany randomization when it is used in greater amounts.

Diffusion Within the reverberator, the Diffusion control can reduce the diffusion provided some of the all pass networks. While the reverb will eventually reach full diffusion regardless of the Diffusion setting, the early reverb diffusion can be reduced, which sometimes is useful to help keep the dry signal "in the clear."

14 Grand Plate

A plate reverberation algorithm.

PAUs: 3

This algorithm emulates an EMT 140 steel plate reverberator. Plate reverberators were manufactured during the 1950's, 1960's, 1970's, and perhaps into the 1980's. By the end of the 1980's, they had been supplanted in the marketplace by digital reverbertors, which first appeared in 1976. While a handful of companies made plate reverberators, EMT (Germany) was the best known and most popular.

A plate reverberator is generally quite heavy and large, perhaps 4 feet high by 7 feet long and a foot thick. They were only slightly adjustable, with controls for high frequency damping and decay time. Some were stereo in, stereo out, others mono in, mono out.

A plate reverb begins with a sheet of plate steel suspended by its edges, leaving the plate free to vibrate. At one (or two) points on the plate, an electromagnetic driver (sort of a small loudspeaker without a cone) is arranged to couple the dry signal into the plate, sending out sound vibrations into the plate in all directions. At one or two other locations, a pickup is placed, sort of like a dynamic microphone whose diaphragm is the plate itself, to pick up the reverberation.

Since the sound waves travel very rapidly in steel (faster than they do in air), and since the dimensions of the plate are not large, the sound quickly reaches the plate edges and reflects from them. This results in a very rapid build up of the reverberation, essentially free of early reflections and with no distinguishable gap before the onset of reverb.

Plates offered a wonderful sound of their own, easily distinguished from other reverberators in the pre-digital reverb era, such as springs or actual "echo" chambers. Plates were bright and diffused (built up echo density) rapidly. Curiously, when we listen to a vintage plate today, we find that the much vaunted brightness is nothing like what we can accomplish digitally; we actually have to deliberately reduce the brightness of a plate emulation to match the sound of a real plate. Similarly, we find that we must throttle back on the low frequency content as well.

The algorithm developed for Grand Plate was carefully crafted for rapid diffusion, low coloration, freedom from discrete early reflections, and "brightness." We also added some controls that were never present in real plates: size, pre delay of up to 500ms, LF damping, low pass roll off, and bass roll off. Furthermore, we allow a wider range of decay time adjustment than a conventional plate. Once the algorithm was complete, we tuned it by presenting the original EMT reverb on one channel and the Grand Plate emulation on the other. A lengthy and careful tuning of Grand Plate (tuning at the micro detail level of each delay and gain in the algorithm) was carried out until the stereo spread of this reverb was matched in all the time periods--early, middle, and late.

The heart of this reverb is the plate simulation network, with its two inputs and two outputs. It is a full stereo reverberation network, which means that the left and right inputs get slightly different treatment in the reverberator. This yields a richer, more natural stereo image from stereo sources. If you have a mono source, assign it to both inputs for best results.

The incoming left source is passed through predelay, low pass (Lowpass), and bass shelf (Bass Gain) blocks. The right source is treated similarly.

There are low pass filters (HF Damping) and high pass filters (LF Damping) embedded in the plate simulation network to modify the decay times. The reverb network also accomodates the Room Size and Decay Time controls.

An output mixer assembles dry and wet signals.

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Room Size	1.00 to 4.00 m		
Pre Dly	0 to 500 ms	Decay Time	0.2 to 5.0 s
HF Damping	16 to 25088 Hz	LF Damping	1 to 294 Hz

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Lowpass	16 to 25088 Hz	Bass Gain	-15 to 0 dB

Wet/Dry The amount of the stereo reverberator (wet) signal relative to the original input (dry) signal sent to the output. The dry signal is not affected by the Lowpass or Bass Gain controls. The wet signal is affected by the Lowpass and Bass Gain controls and by all the other reverberator controls. The balance between wet and dry signals is an extremely important factor in achieving a good mix. Emphasizing the wet signal gives the effect of more reverberation and of greater distance from the source.

Out Gain The overall output level for the reverberation effect and controls the level for both the wet and dry signal paths.

Room Size Choosing an appropriate room size is very important in getting a good reverberation effect. For impulsive sources, such as percussion instruments or plucked strings, increase the size setting until discrete reflections become audible, and then back it off slightly. For slower, softer music, use the largest size possible. At lower settings, Room Size leads to coloration, especially if the Decay Time is set too high. To emulate a plate reverb, this control is typically set to 1.9m.

Pre Dly Introducing predelay creates a gap of silence between the dry sound and the reverberation, allowing the dry signal to stand out with greater clarity and intelligibility against the reverberant background. Especially helpful with vocals or classical music.

Decay Time The reverberation decay time (mid-band "RT60"), the time required before the reverberation has died away to 60dB below its "running" level. Adjust decay time according to the tempo and articulation of the music. To emulate a plate reverb, this control is typically set in the range of 1 to 5 seconds.

HF Damping Adjusts low pass filters in the reverberator so that high frequencies die away more quickly than mid and low frequencies. This shapes the reverberation for a more natural, more acoustically accurate sound. To emulate a plate reverb, a typical value is 5920Hz.

LF Damping Adjusts high pass filters in the reverberator so that low frequencies die away more quickly than mid and high frequencies. This shapes the reverberation for a more natural, more acoustically accurate sound. To emulate a plate reverb, this control is typically set to 52 Hz.

Lowpass Shapes the overall reverberation signal's treble content, but does not modify the decay time. Reduce the treble for a duller, more natural acoustic effect. To emulate a plate reverb, this control is typically set to 3951Hz.

Bass Gain Shapes the overall reverberation signal's bass content, but does not modify the decay time. Reduce the bass for a less muddy sound. To emulate a plate reverb, this control is typically set to -12dB.

15 Finite Verb

Reverse reverberation algorithm.

PAUs: 3

The left and right sources are summed before being fed into a tapped delay line which directly simulates the impulse response of a reverberator. The taps are placed in sequence from zero delay to a maximum delay value, at quasi-regular spacings. By varying the coefficients with which these taps are summed, one can create the effect of a normal rapidly building/slowly decaying reverb or a reverse reverb which builds slowly then stops abruptly.

A special tap is picked off the tapped delay line and its length is controlled by Dly Length. It can be summed into the output wet mix (Dly Lvl) to serve as the simulated dry source that occurs after the reverse reverb sequence has built up and ended. It can also be fed back for special effects. Fdbk Lvl and HF Damping tailor the gain and spectrum of the feedback signal. Despite the complex reverb-like sound of the tapped delay line, the Feedback tap is a pure delay. Feeding it back is like reapplying the source, as in a simple tape echo.

Dly Length and Rvb Length range from 300 to 3000 milliseconds. With the R1 Rvb Env variants, Rvb Length corresponds to a decay time (RT60).

To make things a little more interesting, the tapped delay line mixer is actually broken into three mixers, an early, middle, and late mixer. Each mixes its share of taps and then applies the submix to a low pass filter (cut only) and a simple bass control (boost and cut). Finally, the three equalized sub mixes are mixed into one signal. The Bass and Damp controls allow special effects such as a reverb that begins dull and increases in two steps to a brighter sound.

The Rvb Env control selects 27 cases of envelope gains for the taps. Nine cases emulate a normal forward evolving reverb, but with some special twists. Cases FWD R1xx have a single reverb peak, with a fast attack and slower decay. The sub cases FWD R1Sx vary the sharpness of the envelope, from dullest (S1) to sharpest (S3). The sub cases FWD R2xx have two peaks; that is, the reverb builds, decays, builds again, and decays again. The sub cases FWD R3xx have three peaks.

The sub cases SYM have a symmetrical build and decay time. The cases R1 build to a single peak, while R2 and R3 have two and three peaks, respectively.

The sub cases REV simulate a reverse reverb effect. REV R1xx imitates a backward running reverb, with a long rising "tail" ending abruptly (followed, optionally, by the "dry" source mixed by Dly Lvl). Once again, the number of peaks and the sharpness are variable.

The usual Wet/Dry and Output Gain controls are provided.

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Lvl	0 to 100%		
HF Damping	16 to 25088 Hz		

Page 2

Dly Lvl	0 to 100%	Rvb Env	REV R1S1
Dly Length	300 to 3000 ms	Rvb Length	300 to 3000 ms

Page 3

Early Bass	-15 to 15 dB	Early Damp	16 to 25088 Hz
Mid Bass	-15 to 15 dB	Mid Damp	16 to 25088 Hz
Late Bass	-15 to 15 dB	Late Damp	16 to 25088 Hz

Wet/Dry Wet/Dry sets the relative amount of wet signal and dry signal. The wet signal consistts of the reverb itself (stereo) and the delayed mono signal arriving after the reverb has ended (simulating the dry source in the reverse reverb sequence). The amount of the delayed signal mixed to the Wet signal is separately adjustable with the Dly Lvl control. The Dry signal is the stereo input signal.

Out Gain This controls the level of the output mix, wet and dry, sent back into the K2600.

Fdbk Lvl This controls the feedback gain of the separate, (mono) delay tap. A high value contributes a long repeating echo character to the reverb sound.

HF Damping HF Damping adjusts a low pass filter in the late delay tap feedback path so that high frequencies die away more quickly than mid and low frequencies.

Dly Lvl This adjusts the level of the separate, (mono) delay tap used to simulate the dry source of a reverse reverb effect. This same tap is used for feedback.

Dly Length Sets the length (in milliseconds), of the separate, (mono) delay tap used to simulate the dry source of a reverse reverb effect. This same tap is used for feedback.

Rvb Env The Rvb Env control selects 27 cases of envelope gains for the taps. Nine cases emulate a normal forward evolving reverb, another nine emulate a reverb building symmetrically to a peak at the mid point, while the last nine cases emulate a reverse building reverb. For each major shape, there are three variants of one, two, and three repetitions and three variants of envelope sharpness.

Rvb Length Sets the length (in milliseconds), from start to finish, of the reverberation process. This parameter is essentially the decay time or RT60 for the Rvb Env cases ..R1.. where there is only one repetition.

Bass Early, Mid, and Late. These bass controls shape the frequency response (boost or cut) of the three periods of the finite reverb sequence. Use them to tailor the way the reverb bass content changes with time.

Damp Early, Mid, and Late. These treble controls shape the frequency response (cut only) of the three periods of the finite reverb sequence. Use them to tailor the way the reverb treble content changes with time.

130 Complex Echo

Multitap delay line effect consisting of 6 independent output taps and 4 independent feedback taps

PAUs: 1

Complex Echo is an elaborate delay line with 3 independent output taps per channel, 2 independent feedback taps per channel, equal power output tap panning, feedback diffuser, and high frequency damping. Each channel has three ouptut taps which can each be delayed up to 2600ms (2.6 sec) then panned at the output. Feedback taps can also be delayed up to 2600ms, but both feedback channels do slightly different things. Feedback line 1 feeds the signal back to the delay input of the same channel, while feedback line 2 feeds the signal back to the opposite channel. Feedback line 2 may also be referred to as a "ping-pong" feedback. Relative levels for each feedback line can be set with the "FB2/FB1>FB" control where 0% only allows FB1 to be used, and 100% only allows FB2 to be used.

The diffuser sits at the beginning of the delay line, and consists of three controls. Separate left and right Diff Dly parameters control the length that a signal is smeared from 0 to 100ms as it passes through these diffusers. Diff Amt adjusts the smearing intensity. Short diffuser delays can diffuse the sound while large delays can drastically alter the spectral flavor. Setting all three diffuser parameters to 0 disables the diffuser.

Also at the input to the delays are 1 pole (6dB/oct) lopass filters controlled by the HF Damping parameter.

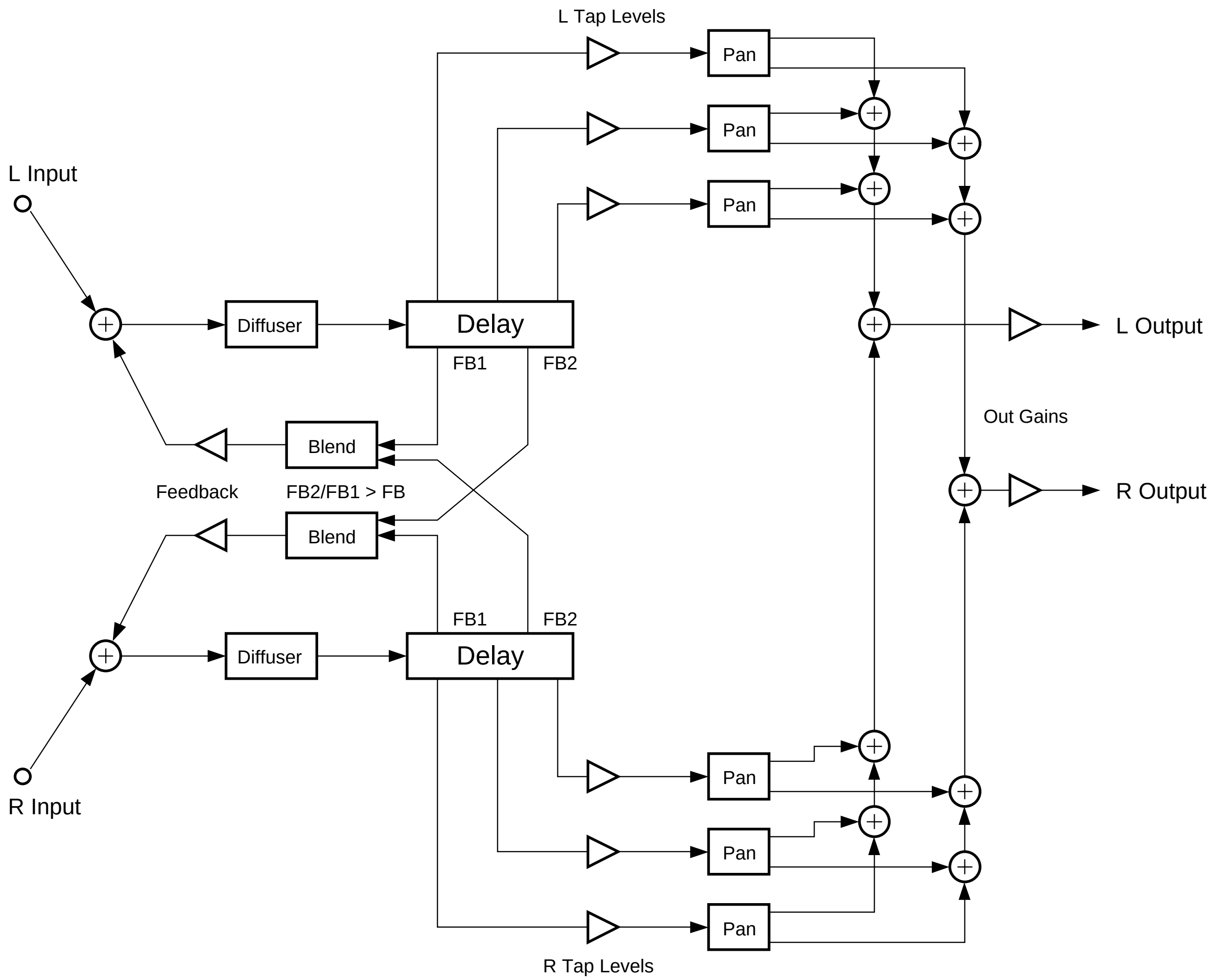


Figure 10-6 Signal flow of Complex Echo

Parameters

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Feedback	0 to 100 %	L Diff Dly	0 to 100 ms
FB2/FB1>FB	0 to 100 %	R Diff Dly	0 to 100 ms
HF Damping	16 to 25088 Hz	Diff Amt	0 to 100 %

Page 2

L Fdbk1 Dly	0 to 2600 ms	R Fdbk1 Dly	0 to 2600 ms
L Fdbk2 Dly	0 to 2600 ms	R Fdbk2 Dly	0 to 2600 ms

L Tap1 Dly	0 to 2600 ms	R Tap1 Dly	0 to 2600 ms
L Tap2 Dly	0 to 2600 ms	R Tap2 Dly	0 to 2600 ms
L Tap3 Dly	0 to 2600 ms	R Tap3 Dly	0 to 2600 ms

Page 3

L Tap1 Lvl	0 to 100 %	R Tap1 Lvl	0 to 100 %
L Tap2 Lvl	0 to 100 %	R Tap2 Lvl	0 to 100 %
L Tap3 Lvl	0 to 100 %	R Tap3 Lvl	0 to 100 %

Page 4

L Tap1 Pan	-100 to 100 %	R Tap1 Pan	-100 to 100 %
L Tap2 Pan	-100 to 100 %	R Tap2 Pan	-100 to 100 %
L Tap3 Pan	-100 to 100 %	R Tap3 Pan	-100 to 100 %

Wet/Dry The relative amount of input signal and effected signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet.

Out Gain The overall gain or amplitude at the output of the effect.

Feedback The amplitude of the feedback tap(s) fed back to the beginning of the delay.

FB2 / FB1>FB Balance control between feedback line 1 and line 2. 0% turns off feedback line 2 only allowing use of feedback line 1. 50% is an even mix of both lines, and 100% turns off line 1.

HF Damping The amount of high frequency content of the signal to the input of the delay. This control determines the cutoff frequency of the one-pole (-6dB/octave) lowpass filters.

Diff Dly Left and Right. Adjusts delay length of the diffusers.

Diff Amt Adjusts the diffuser intensity.

L Fdbk1 Dly Adjusts the delay length of the left channel’s feedback tap fed back to the left channel’s delay input.

L Fdbk2 Dly Adjusts the delay length of the left channel’s feedback tap fed back to the right channel’s delay input.

R Fdbk1 Dly Adjusts the delay length of the right channel’s feedback tap fed back to the right channel’s delay input.

R Fdbk2 Dly Adjusts the delay length of the right channel’s feedback tap fed back to the left channel’s delay input.

Tap*n* Dly Left and Right. Adjusts the delay length of the left and right channel’s three output taps.

Tap*n* Lvl Left and Right. Adjusts the listening level of the left and right channel’s three output taps.

Tap*n* Pan Left and Right. Adjusts the equal power pan position of the left and right channel’s three output taps. 0% is center pan, negative values pan to left, and positive values pan to the right.

131 4-Tap Delay
132 4-Tap Delay BPM

A stereo four tap delay with feedback

PAUs: 1

This is a simple stereo 4 tap delay algorithm with delay lengths defined in milliseconds (ms). The left and right channels are fully symetric (all controls affect both channels). The duration of each stereo delay tap (length of the delay) and the signal level from each stereo tap may be set. Prior to output each delay tap passes through a level and left-right balance control. The taps are summed and added to the dry input signal through a Wet/Dry control. The delayed signal from the "Loop" tap may be fed back to the delay input.

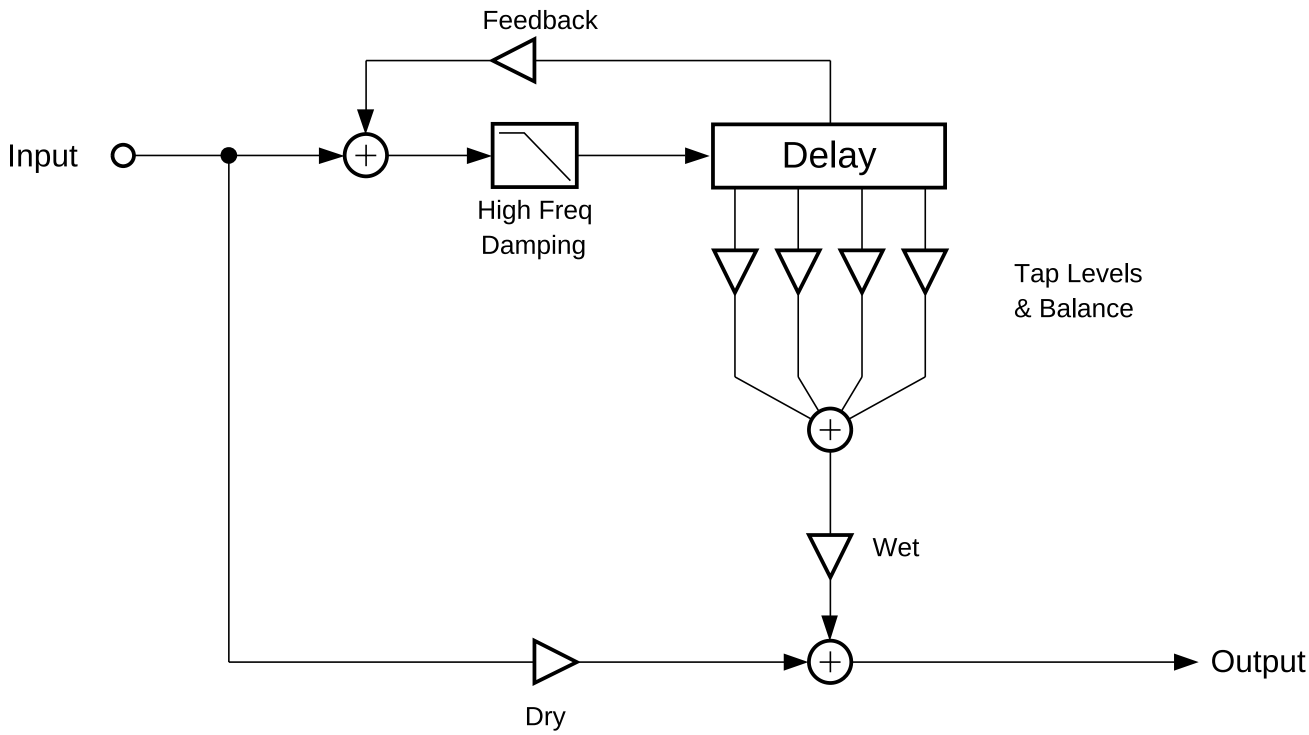


Figure 10-7 Left Channel of 4-Tap Delay

The delay length for any given tap is the sum of the coarse and fine parameters for the tap multiplied by the DelayScale parameter which is common to all taps. The DelayScale parameter allows you to change the lengths of all the taps together.

A repetitive loop delay is created by turning up the Fdbk Level parameter. Only the Loop tap is fed back to the input of the delay, so this is the tap which controls the loop rate. Usually you will want the Loop delay length to be longer than the other tap lengths. Set the Loop delay length to the desired length then set the other taps to fill in the measure with interesting rhythmical patterns. Setting tap levels allows some "beats" to receive different emphasis than others. The delay lengths for 4-Tap Delay are in units of milliseconds (ms). If you want to base delay lengths on tempo, then the 4-Tap Delay BPM algorithm may be more convenient.

The feedback (Fdbk Level) controls how long a sound in the delay line takes to die out. At 100% feedback, your sound will be repeated indefinitely. HF Damping selectively removes high frequency content from your delayed signal and will also cause your sound to eventually disappear.

The Hold parameter is a switch which controls signal routing. When turned on, Hold will play whatever signal is in the delay line indefinitely. Hold overrides the feedback parameter and prevents any incoming signal from entering the delay. You may have to practice using the Hold parameter. Each time your sound goes through the delay, it is reduced by the feedback amount. If feedback is fairly low and you turn on Hold at the wrong moment, you can get a disconcerting jump in level at some point in the loop. The Hold parameter has no effect on the Wet/Dry or HF Damping parameters, which continue to work as usual, so if there is some HF Damping, the delay will eventually die out.

See also the versions of these algorithms which specify delay lengths in terms of tempo and beats.

Parameters for Algorithm 131 4-Tap Delay

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	0 to 100%		
		Dry Bal	-100 to 100%
HF Damping	16 Hz to 25088 Hz	Hold	On or Off

Page 2

Loop Crs	0 to 2540 ms	DelayScale	0.00x to 10.00x
Loop Fine	-20 to 20 ms		
Tap1 Crs	0 to 2540 ms	Tap3 Crs	0 to 2540 ms
Tap1 Fine	-20 to 20 ms	Tap3 Fine	-20 to 20 ms
Tap2 Crs	0 to 2540 ms	Tap4 Crs	0 to 2540 ms
Tap2 Fine	-20 to 20 ms	Tap4 Fine	-20 to 20 ms

Page 3

Loop Level	0 to 100 %	Loop Bal	-100 to 100 %
Tap2 Level	0 to 100 %	Tap2 Bal	-100 to 100 %
Tap3 Level	0 to 100 %	Tap3 Bal	-100 to 100 %
Tap4 Level	0 to 100 %	Tap4 Bal	-100 to 100 %

Wet/Dry The relative amount of input signal and delay signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet.

Out Gain The overall gain or amplitude at the output of the effect.

Fdbk Level The percentage of the delayed signal to feed back or return to the delay input. Turning up the feedback will cause the effect to repeatedly echo or act as a crude reverb.

HF Damping The -3 dB frequency in Hz of a one pole lowpass filter (-6 dB/octave) placed in front of the delay line. The filter is specified for a signal passing through the filter once. Multiple passes through the feedback will cause the signal to become more and more dull.

Dry Bal The left-right balance of the dry signal. A setting of -100% allows only the left dry signal to pass to the left output, while a setting of 100% lets only the right dry signal pass to the right output. At 0%, equal amounts of the left and right dry signals pass to their respective outputs.

Hold A switch which when turned on, locks any signal currently in the delay to play until Hold is turned off. When Hold is on, no signal can enter the delay and Feedback is set to 100% behind the scenes. Hold does not affect the HF Damping or Wet/Dry mix.

Loop Crs The coarse delay length of the Loop tap. If the feedback is turned up, this parameter sets the repeating delay loop length. The resolution of the coarse adjust is 20 milliseconds, but finer resolution can be obtained using the Loop Fine parameter. The maximum delay length is 2.55 seconds (2550ms) for the 4-Tap Delay.

Loop Fine A fine adjustment to the Loop tap delay length. The delay resolution is 0.2 milliseconds (ms). Loop Fine is added to Loop Crs (coarse) to get the actual delay length.

Tapn Crs The coarse delay lengths of the output taps (n = 1...4). The resolution of the coarse adjust is 20 milliseconds, but finer resolution can be obtained using the Tapn Fine parameters. The maximum delay length is 2.55 seconds (2550ms) for the 4-Tap Delay.

Tap*n* Fine A fine adjustment to the output tap delay lengths (n = 1...4). The delay resolution is 0.2 milliseconds (ms). Tapn Fine is added to Tapn Crs (coarse) to get actual delay lengths.

Tap*n* Level The amount of signal from each of the taps (n = 1...4) which get sent to the output. With the Loop Lvl control, you can give different amounts of emphasis to various taps in the loop.

Tap*n* Bal The left-right balance of each of the stereo taps (n = 1...4). A setting of -100% allows only the left tap to pass to the left output, while a setting of 100% lets only the right tap pass to the right output. At 0%, equal amounts of the left and right taps pass to their respective outputs.

Algorithm 132 4-Tap Delay BPM

In this Algorithm, the delay length for any given tap is determined by the tempo, expressed in beats per minute (BPM), and the delay length of the tap expressed in beats (bts). The tempo alters all tap lengths together. With the tempo in beats per minute and delay lengths in beats, you can calculate the length of a delay in seconds as beats/tempo * 60 (sec/min). IMPORTANT NOTE: KDFX has a limited amount of delay memory available (over 2.5 seconds for 4-Tap BPM). When slow tempos and/or long lengths are specified, you may run out of delay memory, at which point the delay length will be cut in half. When you slow down the tempo, you may find the delays suddenly getting shorter.

A repetitive loop delay is created by turning up the feedback parameter (Fdbk Level). Only the Loop tap is fed back to the input of the delay, so this is the tap which controls the loop rate. Usually you will want the Loop tap (LoopLength parameter) to be longer than the other tap lengths. To repeat a pattern on a 4/4 measure (4 beats per measure) simply set LoopLength to 4 bts. The output taps can then be used to fill in the measure with interesting rhythmical patterns. Setting tap levels allows some “beats” to receive different emphasis than others.

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	0 to 100%	Tempo	System, 1 to 255 BPM
		Dry Bal	-100 to 100%
HF Damping	16 Hz to 25088 Hz	Hold	On or Off

Page 2

LoopLength	0 to 32 bts		
Tap1 Delay	0 to 32 bts		
Tap2 Delay	0 to 32 bts		
Tap3 Delay	0 to 32 bts		
Tap4 Delay	0 to 32 bts		

Page 3

Tap1 Level	0 to 100 %	Tap1 Bal	-100 to 100 %
Tap2 Level	0 to 100 %	Tap2 Bal	-100 to 100 %
Tap3 Level	0 to 100 %	Tap3 Bal	-100 to 100 %
Tap4 Level	0 to 100 %	Tap4 Bal	-100 to 100 %

Tempo Basis for the delay lengths, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LoopLength The delay length of the Loop tap. If the feedback is turned up, this parameter sets the repeating delay loop length. LoopLength sets the loop delay length as a tempo beat duration. The tempo is specified with the Tempo parameter and the delay length is given in beats (bts). The delay length in seconds is calculated as beats/tempo * 60 (sec/min).

Tap*n* Delay The delay lengths of the taps (n = 1...4) as tempo beat durations. The tempo is specified with the Tempo parameter and the delay length is given in beats (bts). The delay length in seconds is calculated as beats/tempo * 60 (sec/min). Use the output taps to create interesting rhythmic patterns within the repeating loop.

133 8-Tap Delay
134 8-Tap Delay BPM

A stereo eight tap delay with cross-coupled feedback

PAUs: 2

This is a simple stereo 8 tap delay algorithm with delay lengths defined in milliseconds (ms). The left and right channels are fully symmetric (all controls affect both channels). The duration of each stereo delay tap (length of the delay) and the signal level from each stereo tap may be set. Prior to output each delay tap passes through a level and left-right balance control. Pairs of stereo taps are tied together with balance controls acting with opposite left-right sense. The taps are summed and added to the dry input signal throught a Wet/Dry control. The delayed signal from the “Loop” tap may be fed back to the delay input. The sum of the input signal and the feedback signal may be mixed or swapped with the input/feedback signal from the other channel (cross-coupling). When used with feedback, cross-coupling can achieve a ping-pong effect between the left and right channels.

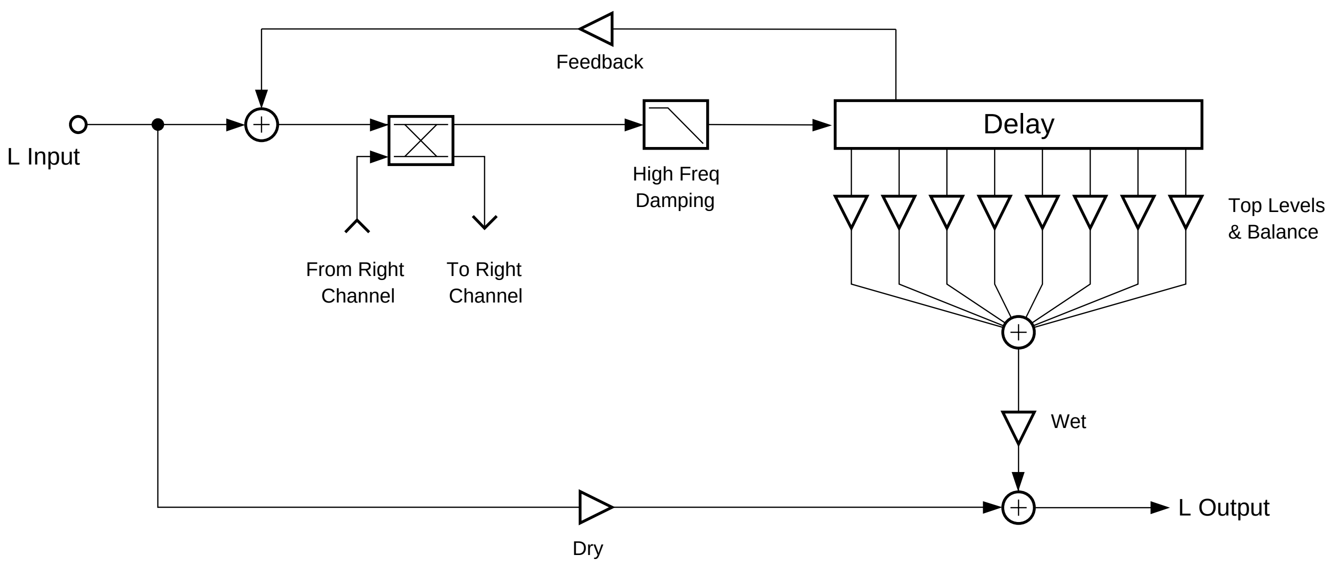


Figure 10-8 Left Channel of 8-Tap Delay

The delay length for any given tap is the sum of the coarse and fine parameters for the tap multiplied by the DelayScale parameter which is common to all taps. The DelayScale parameter allows you to change the lengths of all the taps together.

A repetitive loop delay is created by turning up the Fdbk Level parameter. Only the Loop tap is fed back to the input of the delay, so this is the tap which controls the loop rate. Usually you will want the Loop delay length to be longer than the other tap lengths. Set the Loop delay length to the desired length then set the other taps to fill in the measure with interesting rhythmical patterns. Setting tap levels allows some “beats” to receive different emphasis than others. The delay lengths for 8-Tap Delay are in units of milliseconds (ms). If you want to base delay lengths on tempo, then the 8-Tap Delay BPM algorithm may be more convenient.

The feedback (Fdbk Level) controls how long a sound in the delay line takes to die out. At 100% feedback, your sound will be repeated indefinitely. HF Damping selectively removes high frequency content from your delayed signal and will also cause your sound to eventually disappear.

The Hold parameter is a switch which controls signal routing. When turned on, Hold will play whatever signal is in the delay line indefinitely. Hold overrides the feedback parameter and prevents any incoming signal from entering the delay. You may have to practice using the Hold parameter. Each time your sound goes through the delay, it is reduced by the feedback amount. If feedback is fairly low and you turn on Hold at the wrong moment, you can get a disconcerting jump in level at some point in the loop. The Hold parameter has no effect on the Wet/Dry or HF Damping parameters, which continue to work as usual, so if there is some HF Damping, the delay will eventually die out.

See also the versions of these algorithms which specify delay lengths in terms of tempo and beats.

Parameters for Algorithm 133 8-Tap Delay

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	0 to 100%		
Xcouple	0 to 100%	Dry Bal	-100 to 100%
HF Damping	16 Hz to 25088 Hz	Hold	On or Off

Page 2

Loop Crs	0 to 5100 ms	DelayScale	0.00x to 10.00x
Loop Fine	-20 to 20 ms		
Tap1 Crs	0 to 5100 ms	Tap3 Crs	0 to 5100 ms
Tap1 Fine	-20 to 20 ms	Tap3 Fine	-20 to 20 ms
Tap2 Crs	0 to 5100 ms	Tap4 Crs	0 to 5100 ms
Tap2 Fine	-20 to 20 ms	Tap4 Fine	-20 to 20 ms

Page 3

Tap5 Crs	0 to 5100 ms	Tap7 Crs	0 to 5100 ms
Tap5 Fine	-20 to 20 ms	Tap7 Fine	-20 to 20 ms
Tap6 Crs	0 to 5100 ms	Tap8 Crs	0 to 5100 ms
Tap6 Fine	-20 to 20 ms	Tap8 Fine	-20 to 20 ms

Page 4

Tap1 Level	0 to 100 %	Tap5 Level	0 to 100 %
Tap2 Level	0 to 100 %	Tap6 Level	0 to 100 %
Tap3 Level	0 to 100 %	Tap7 Level	0 to 100 %
Tap4 Level	0 to 100 %	Tap8 Level	0 to 100 %
Tap1/-5Bal	-100 to 100 %	Tap3/-7Bal	-100 to 100 %
Tap2/-6Bal	-100 to 100 %	Tap4/-8Bal	-100 to 100 %

Wet/Dry The relative amount of input signal and delay signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet.

Out Gain The overall gain or amplitude at the output of the effect.

Fdbk Level The percentage of the delayed signal to feed back or return to the delay input. Turning up the feedback will cause the effect to repeatedly echo or act as a crude reverb.

Xcouple 8 Tap Delay is a stereo effect. The cross coupling control lets you send the feedback from a channel to its own input (0% cross coupling) or to the other channel's input (100% cross coupling) or somewhere in between. This control has no effect if the Fdbk Level control is set to 0%.

HF Damping The -3 dB frequency in Hz of a one pole lowpass filter (-6 dB/octave) placed in front of the delay line. The filter is specified for a signal passing through the filter once. Multiple passes through the feedback will cause the signal to become more and more dull.

Dry Bal The left-right balance of the dry signal. A setting of -100% allows only the left dry signal to pass to the left output, while a setting of 100% lets only the right dry signal pass to the right output. At 0%, equal amounts of the left and right dry signals pass to their respective outputs.

Hold A switch which when turned on, locks any signal currently in the delay to play until Hold is turned off. When Hold is on, no signal can enter the delay and Feedback is set to 100% behind the scenes. Hold does not affect the HF Damping or Wet/Dry mix.

Loop Crs The coarse delay length of the Loop tap. If the feedback is turned up, this parameter sets the repeating delay loop length. The resolution of the coarse adjust is 20 milliseconds, but finer resolution can be obtained using the Loop Fine parameter. The maximum delay length is 5.10 seconds (5100ms) for the 8-Tap Delay.

Loop Fine A fine adjustment to the Loop tap delay length. The delay resolution is 0.2 milliseconds (ms). Loop Fine is added to Loop Crs (coarse) to get the actual delay length.

Tapn **Crs** The coarse delay lengths of the output taps (*n* = 1...8). The resolution of the coarse adjust is 20 milliseconds, but finer resolution can be obtained using the Tap*n* Fine parameters. The maximum delay length is 5.1 seconds (5100ms) for the 8-Tap Delay.

Tapn **Fine** A fine adjustment to the output tap delay lengths (*n* = 1...8). The delay resolution is 0.2 milliseconds (ms). Tap*n* Fine is added to Tap*n* Crs (coarse) to get actual delay lengths.

Tapn **Leve**l The amount of signal from each of the taps (*n* = 1...8) which get sent to the output.

Tap*m*/-*n* Bal The left-right balance of each of the stereo taps. The balances are controlled in pairs of taps: 1 & 5, 2 & 6, 3 & 7, and 4 & 8. The balance controls work in opposite directions for the two taps in the pair. When the balance is set to -100%, the first tap of the pair is fully right while the second is fully left. At 0%, equal amounts of the left and right taps pass to their respective outputs.

Algorithm 134: 8-Tap Delay BPM

In this Algorithm the delay length for any given tap is determined by the tempo, expressed in beats per minute (BPM), and the delay length of the tap expressed in beats (bts). The tempo alters all tap lengths together. With the tempo in beats per minute and delay lengths in beats, you can calculate the length of a delay in seconds as beats/tempo * 60 (sec/min). IMPORTANT NOTE: KDFX has a limited amount of delay memory available (over 5 seconds for 8 Tap Delay BPM). When slow tempos and/or long lengths are specified, you may run out of delay memory, at which point the delay length will be cut in half. When you slow down the tempo, you may find the delays suddenly getting shorter.

A repetitive loop delay is created by turning up the feedback parameter (Fdbk Level). Only the Loop tap is fed back to the input of the delay, so this is the tap which controls the loop rate. Usually you will want the Loop tap (LoopLength parameter) to be longer than the other tap lengths. To repeat a pattern on a 4/4 measure (4 beats per measure) simply set LoopLength to 4 bts. The output taps can then be used to fill in the measure with interesting rhythmical patterns. Setting tap levels allows some "beats" to receive different emphasis than others.

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	0 to 100%	Tempo	System, 1 to 255 BPM
Xcouple	0 to 100%	Dry Bal	-100 to 100%
HF Damping	16 Hz to 25088 Hz	Hold	On or Off

Page 2

LoopLength	0 to 32 bts		
Tap1 Delay	0 to 32 bts	Tap5 Delay	0 to 32 bts
Tap2 Delay	0 to 32 bts	Tap6 Delay	0 to 32 bts
Tap3 Delay	0 to 32 bts	Tap7 Delay	0 to 32 bts
Tap4 Delay	0 to 32 bts	Tap8 Delay	0 to 32 bts

Page 3

Tap1 Level	0 to 100 %	Tap5 Level	0 to 100 %
Tap2 Level	0 to 100 %	Tap6 Level	0 to 100 %
Tap3 Level	0 to 100 %	Tap7 Level	0 to 100 %
Tap4 Level	0 to 100 %	Tap8 Level	0 to 100 %

Page 4

Tap1 Bal	-100 to 100 %	Tap5 Bal	-100 to 100 %
Tap2 Bal	-100 to 100 %	Tap6 Bal	-100 to 100 %
Tap3 Bal	-100 to 100 %	Tap7 Bal	-100 to 100 %
Tap4 Bal	-100 to 100 %	Tap8 Bal	-100 to 100 %

Tempo Basis for the delay lengths, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to "System", the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to "System", sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LoopLength The delay length of the Loop tap. If the feedback is turned up, this parameter sets the repeating delay loop length. LoopLength sets the loop delay length as a tempo beat duration. The tempo is specified with the Tempo parameter and the delay length is given in beats (bts). The delay length in seconds is calculated as beats/tempo * 60 (sec/min).

Tap*n* Delay The delay lengths of the taps (n = 1...8) as tempo beat durations. The tempo is specified with the Tempo parameter and the delay length is given in beats (bts). The delay length in seconds is calculated as beats/tempo * 60 (sec/min). Use the output taps to create interesting rhythmic patterns within the repeating loop.

135 Spectral 4-Tap
136 Spectral 6-Tap

Tempo based 4 and 6 tap delays with added shapers and resonant comb filters on each tap

PAUs: 2 for Spectral 4-Tap
3 for Spectral 6-Tap

Spectral 4 Tap and Spectral 6 Tap are respectively 2 and 3 processing allocation unit (PAU) tempo based multi-tap delay effects. They are similar to a simple 4 and 6 tap delays with feedback, but have their feedback and output taps modified with shapers and filters. In the feedback path of each are a diffuser, hipass filter, lopass filter, and imager. Each delay tap has a shaper, comb filter, balance and level controls with the exception of Tap 1, which does not have a comb filter (Figure 1).

Diffusers add a quality that can be described as “smearing” the feedback signal. The more a signal has been regenerated through feedback and consequently fed through the diffuser, the more it is smeared. It requires two parameters, one for the duration a signal is smeared labeled Diff Delay, and the other for the amount it is smeared labeled Diff Amt. Positive diffusion settings will add diffusion while maintaining image integrity. Negative diffusion amounts will cause the feedback image to lose image integrity and become wide. Short Diff Delay settings have subtle smearing effects. Increasing Diff Delay will be more noticeable, and long delay settings will take on a ringy resonant quality. To disable the diffuser, both Diff Delay and Diff Amt should be set to zero.

Two 1 pole 6dB/oct filters are also in the feedback path: hipass and lopass. The hipass filter roll-off frequency is controlled with LF Damping, and the lopass filter roll-off frequency is controlled by HF Damping.

The imager (found on PARAM2) shifts the stereo input image when fed through feedback. Small positive or negative values shift the image to the right or left respectively. Larger values shift the image so much that the image gets scrambled through each feedback generation.

On each output tap is a shaper. For an overview of shaper functionality, refer to the section on shapers in the *Musician’s Guide*. The Spectral Multi-Tap shapers offer 4 shaping loops as opposed to 8 found in the VAST shapers, but can allow up to 6.00x intensity (Figure 2). Immediately following the shapers on taps 2 and above are resonant comb filters tuned in semitones. These comb filters make the taps become pitched. When a comb filter is in use, the shaper before it can be used to intensify these pitched qualities.

Each tap also has separate balance and level controls.

Since these are tempo based effects, tap delay values and feedback delay (labeled LoopLength on PARAM2) values are set relative to a beat. The beat duration is set be adjusting Tempo in BPM. The tempo can be synced to the system clock by setting Tempo to System. Each tap’s delay is adjusted relative to 1 beat, in 1/24 beat increments. Notice that 24 is a musically useful beat division because it can divide a beat into halves, 3rds, 4ths, 6ths, 8ths, 12ths, and of course 24ths. For example, setting LoopLength to “1 12/24bts” will put the feedback tap at 1 1/2 beats (dotted quarter note in 4/4 time) of delay making the feedback repetition occur every one and a half beats. This is equivalent to 3/4 of a second at 120 BPM.

When Temp is set to 60 BPM, each 1/24th of a beat is equivalent to 1/24th of a second. When tempo is set to 250 BPM, each 1/24th of a beat is equivalent to 10ms of delay.

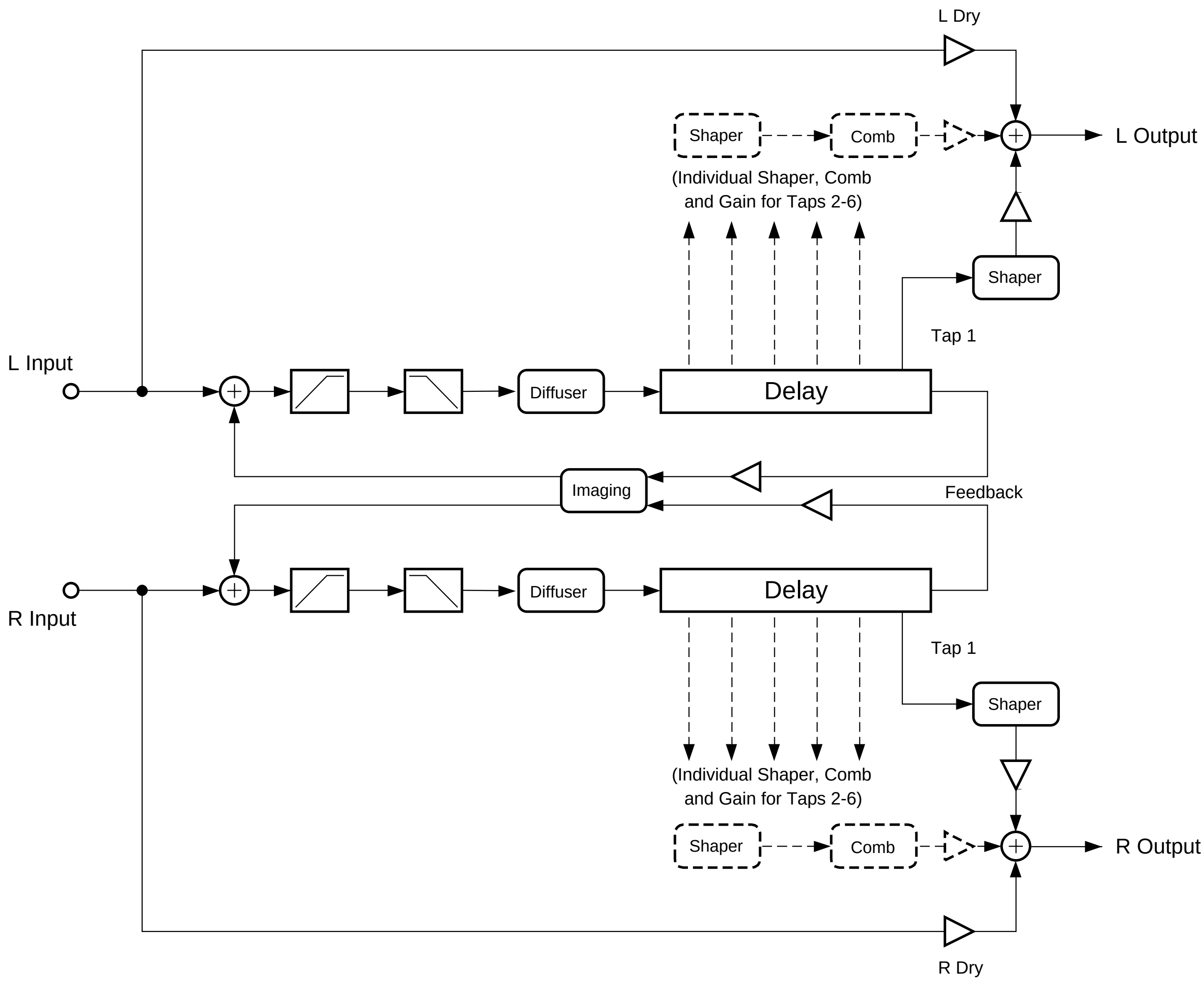


Figure 10-9 Spectral 6 Tap

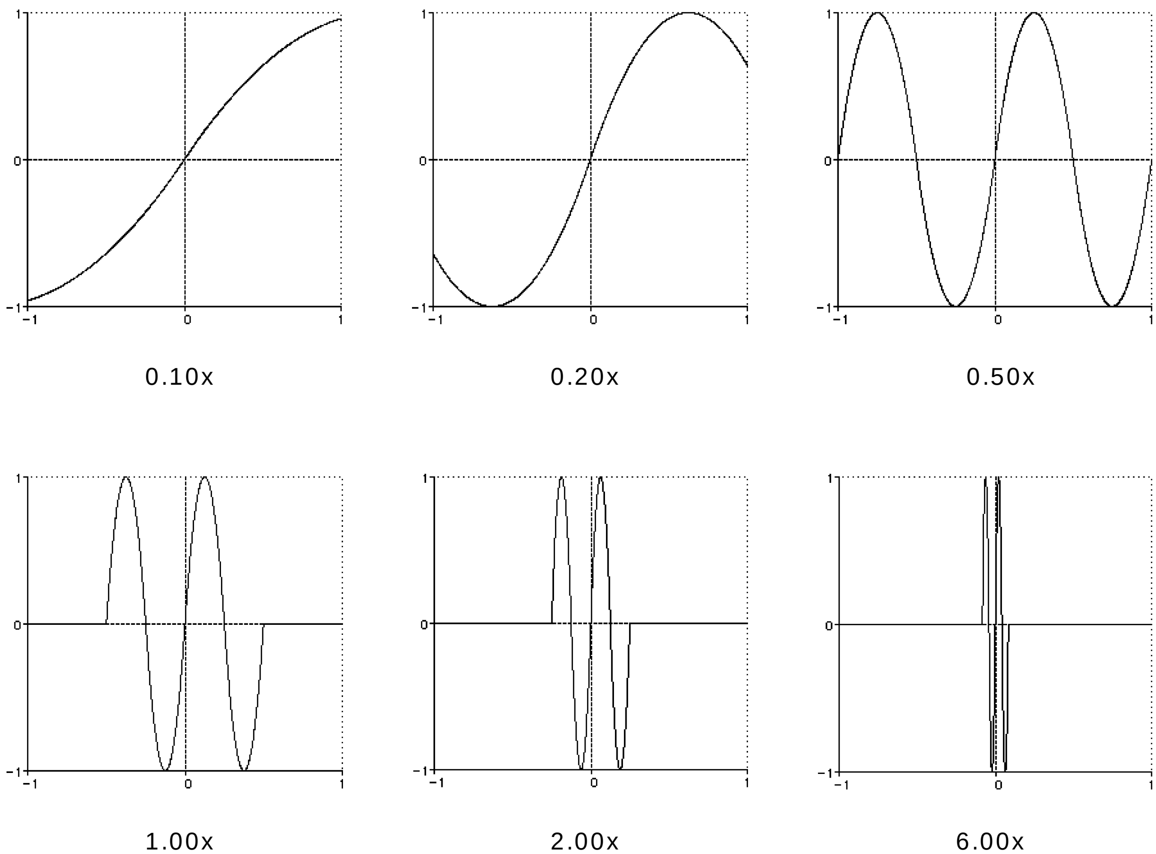


Figure 10-10 Various shaper curves used in the Spectral Multi-Taps

Parameters for Spectral 4-Tap

Page 1

Wet/Dry	0 to 100 %	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	0 to 100 %	Tempo	System, 0 to 255 BPM
HF Damping	16 to 25088 Hz	Diff Delay	0 to 20.0 ms
LF Damping	16 to 25088 Hz	Diff Amt	-100 to 100 %

Page 2

LoopLength	On or Off	Tap2 Delay	0 to 32 bts
Fdbk Image	-100 to 100 %	Tap2 Shapr	0.10 to 6.00 x
Tap1 Delay	0 to 32 bts	Tap2 Pitch	C-1 to C8
Tap1 Shapr	0.10 to 6.00 x	Tap2 PtAmt	0 to 100%
Tap1 Level	0 to 100 %	Tap2 Level	0 to 100%
Tap1 Bal	-100 to 100 %	Tap2 Bal	-100 to 100%

Page 3

Tap3 Delay	0 to 32 bts	Tap4 Delay	0 to 32 bts
Tap3 Shapr	0.10 to 6.00 x	Tap4 Shapr	0.10 to 6.00 x
Tap3 Pitch	C-1 to C8	Tap4 Pitch	C-1 to C8
Tap3 PtAmt	0 to 100%	Tap4 PtAmt	0 to 100%
Tap3 Level	0 to 100%	Tap4 Level	0 to 100%
Tap3 Bal	-100 to 100%	Tap4 Bal	-100 to 100%

Parameters for Spectral 6-Tap

Page 1

Wet/Dry	0 to 100 %	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	0 to 100 %	Tempo	System, 0 to 255 BPM
HF Damping	16 to 25088 Hz	Diff Delay	0 to 20.0 ms
LF Damping	16 to 25088 Hz	Diff Amt	-100 to 100 %

Page 2

LoopLength	On or Off	Tap2 Delay	0 to 32 bts
Fdbk Image	-100 to 100 %	Tap2 Shapr	0.10 to 6.00 x
Tap1 Delay	0 to 32 bts	Tap2 Pitch	C-1 to C8
Tap1 Shapr	0.10 to 6.00 x	Tap2 PtAmt	0 to 100%
Tap1 Level	0 to 100 %	Tap2 Level	0 to 100%
Tap1 Bal	-100 to 100 %	Tap2 Bal	-100 to 100%

Page 3

Tap3 Delay	0 to 32 bts	Tap4 Delay	0 to 32 bts
Tap3 Shapr	0.10 to 6.00 x	Tap4 Shapr	0.10 to 6.00 x
Tap3 Pitch	C-1 to C8	Tap4 Pitch	C-1 to C8
Tap3 PtAmt	0 to 100%	Tap4 PtAmt	0 to 100%
Tap3 Level	0 to 100%	Tap4 Level	0 to 100%
Tap3 Bal	-100 to 100%	Tap4 Bal	-100 to 100%

Page 4

Tap5 Delay	0 to 32 bts	Tap6 Delay	0 to 32 bts
Tap5 Shapr	0.10 to 6.00 x	Tap6 Shapr	0.10 to 6.00 x
Tap5 Pitch	C-1 to C8	Tap6 Pitch	C-1 to C8
Tap5 PtAmt	0 to 100%	Tap6 PtAmt	0 to 100%
Tap5 Level	0 to 100%	Tap6 Level	0 to 100%
Tap5 Bal	-100 to 100%	Tap6 Bal	-100 to 100%

Wet/Dry	The relative amount of input signal and effected signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet. Negative values polarity invert the wet signal.
Out Gain	The overall gain or amplitude at the output of the effect.
Fdbk Level	The amount that the feedback tap is fed to the input of the delay.
HF Damping	The amount of high frequency content of the signal to the input of the delay. This control determines the cutoff frequency of the one-pole (-6dB/octave) lopass filters.
LF Damping	The amount of low frequency content of the signal to the input of the delay. This control determines the cutoff frequency of the one-pole (-6dB/octave) lopass filters.
Tempo	Basis for the rates of the delay times, as referenced to a musical tempo in BPM (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.
Diff Dly	The length that the diffuser smears the signal sent to the input of the delay.
Diff Amt	The intensity that the diffuser smears the signal sent to the input of the delay. Negative values decorrelate the stereo signal.
LoopLength	The delay length of the feedback tap in 24ths of a beat.
Fdbk Image	Sets the amount the stereo image is shifted each time it passes through the feedback line.
Tap *n* Delay	Adjusts the length of time in 24ths of a beat each output tap is delayed.
Tap *n* Shapr	Adjusts the intensity of the shaper at each output tap.
Tap *n* Pitch	Adjusts the frequency in semitones of the comb filter at each output tap.
Tap *n* PtAmt	Adjusts the intensity of the comb filter at each output tap.
Tap *n* Level	Adjusts the relative amplitude that each output tap is heard.
Tap *n* Bal	Adjusts the left/right balance of each output tap. Negative values bring down the right channel, and positive values bring down the left channel.

Algorithms 150–153: Choruses

150 Chorus 1
151 Chorus 2
152 Dual Chorus 1
153 Dual Chorus 2

One and three tap dual mono choruses

PAUs: 1 for Chorus 1 (both)
2 for Chorus 2 (both)

Chorus is an effect that gives the illusion of multiple voices playing in unison. The effect is achieved by detuning copies of the original signal and summing the detuned copies back with the original. Low frequency oscillators (LFOs) are used modulate the positions of output taps from a delay line. The delay line tap modulation causes the pitch of the signal to shift up and down, producing the required detuning.

The choruses are available as stereo or dual mono. The stereo choruses have the parameters for the left and right channels ganged.

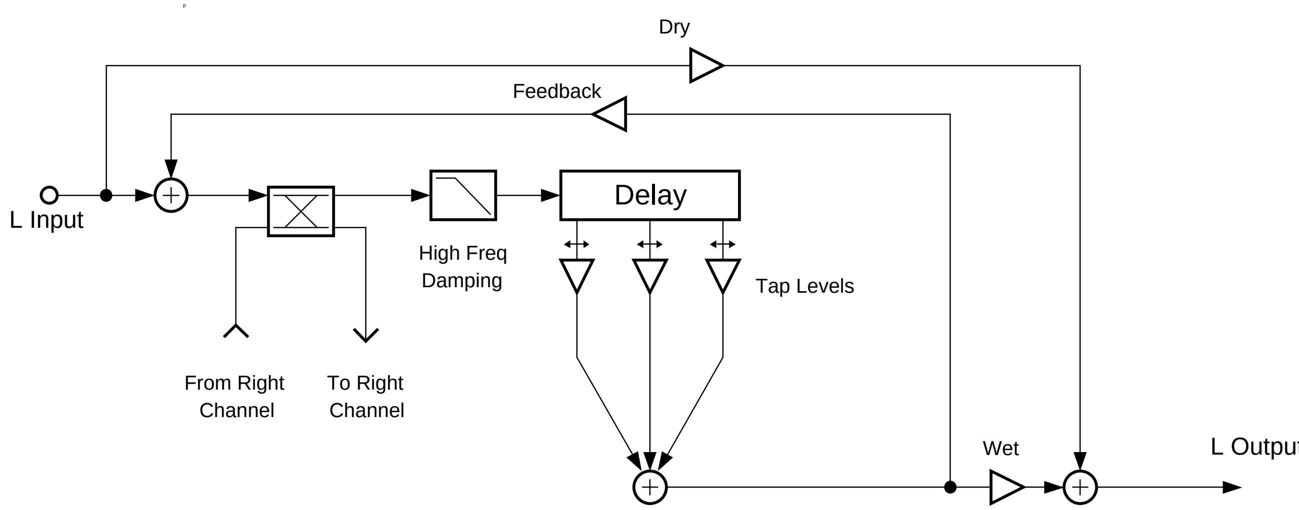


Figure 10-11 Block diagram of left channel of Chorus 2

Right channel is the same.

Chorus 2 is a 2 unit allocation multi-tapped delay (3 taps) based chorus effect with cross-coupling and individual output tap panning. Figure 10-11 is a simplified block diagram of the left channel of Chorus 2.

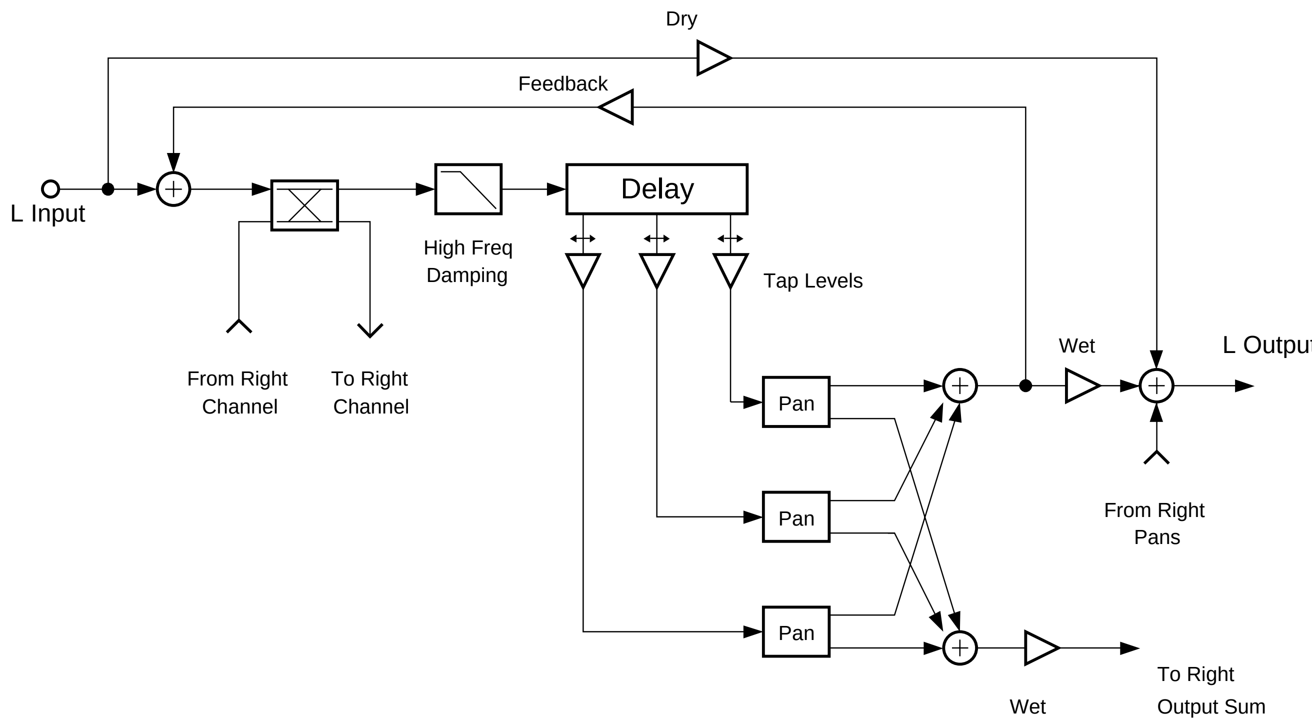


Figure 10-12 Block Diagram of Left Channel of Dual Chorus 2 (right channel is similar)

The dual mono choruses are like the stereo choruses but have separate left and right controls. Dual mono choruses also allow you to pan the delay taps between left or right outputs

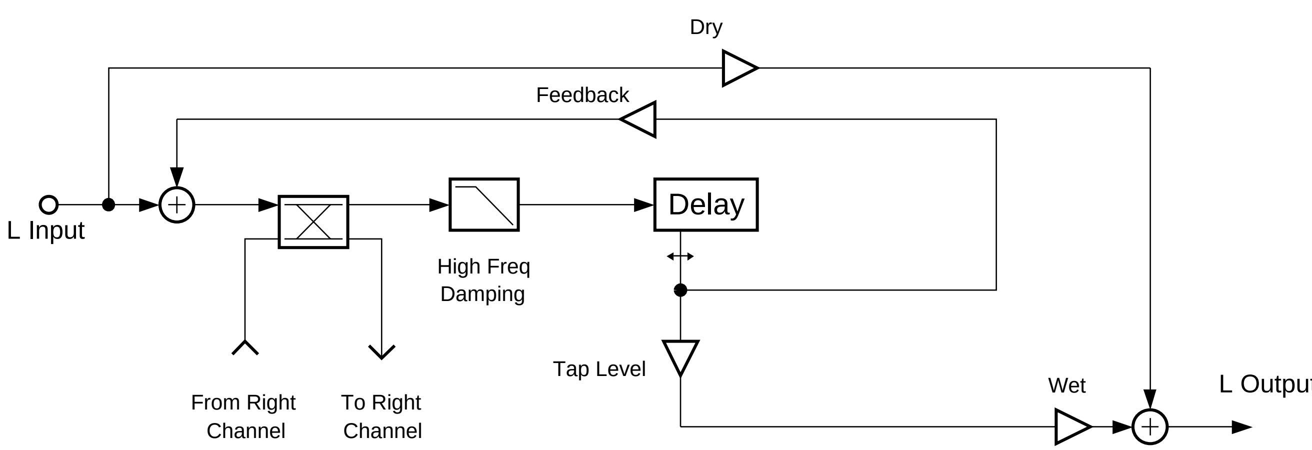


Figure 10-13 Block diagram of left channel of Chorus 1 (right channel is the same)

Chorus 1 uses just 1 unit allocation and has one delay tap. Figure 10-13 is a simplified block diagram of the left channel of Chorus 1.

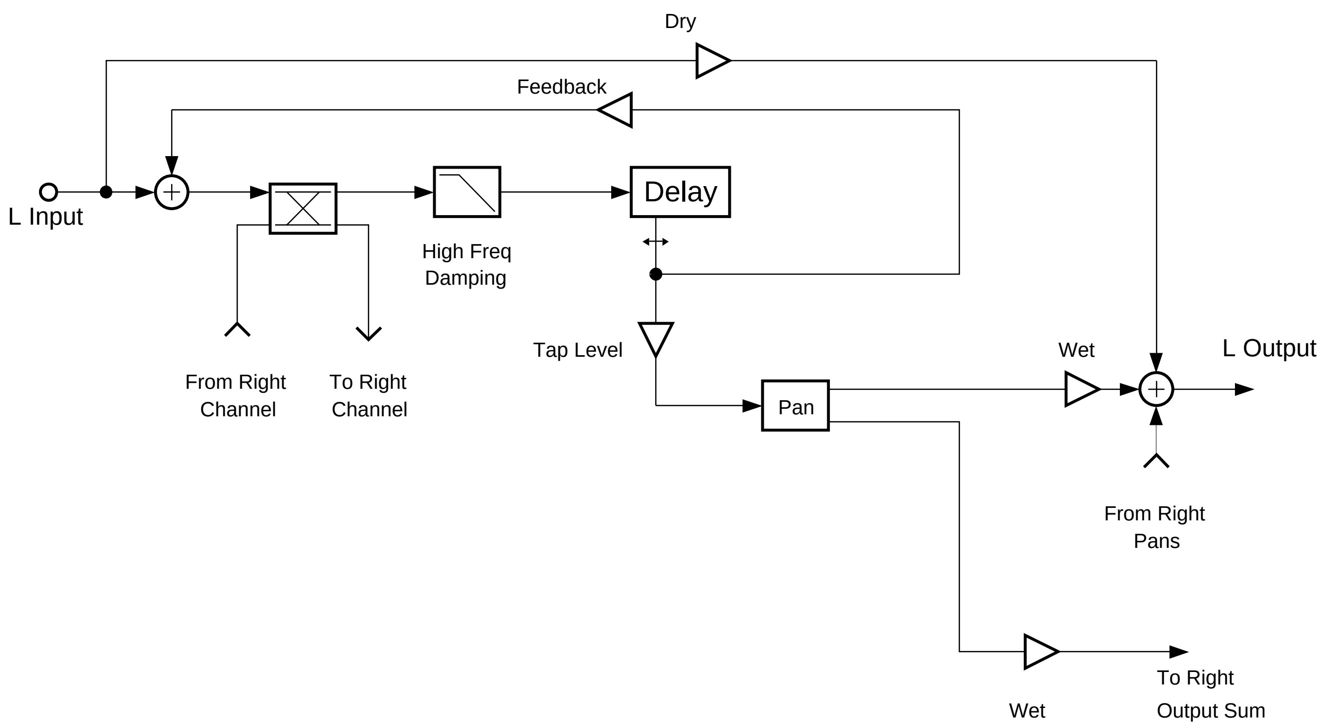


Figure 10-14 Block diagram of left channel of Dual Chorus 1 (right channel is similar)

The left and right channels pass through their own chorus blocks and there may be cross-coupling between the channels. For Chorus 2 and Dual Chorus 2, each channel has three moving taps which are summed, while Chorus 1 and Dual Chorus 2 have one moving tap for both channels. For the dual mono choruses you can pan the taps to left or right. The summed taps (or the single tap of Chorus 1) is used for the wet output signal. The summed tap outputs, weighted by their level controls, are used for feedback back to the delay line input. The input and feedback signals go through a one pole lowpass filter (HF Damping) before going entering the delay line.

The Wet/Dry control is an equal power crossfade. Note that the Output Gain parameters affects both wet and dry signals.

For each of the LFO tapped delay lines, you may set the tap levels, the left/right pan position, delays of the modulating delay lines, the rates of the LFO cycles, and the maximum depths of the pitch detuning. The LFOs detune the pitch of signal copies above *and* below the original pitch. The depth units are in cents, and there are 100 cents in a semitone.

In the stereo Chorus 1 and Chorus 2, the relative phases of the LFOs modulating the left and right channels may be adjusted.

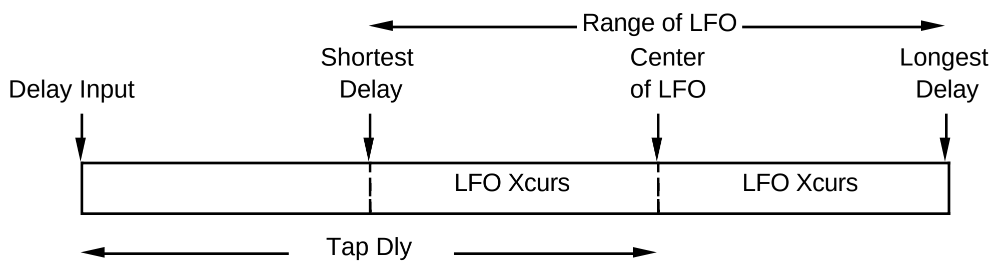


Figure 10-15 Delay for a Single LFO

The settings of the LFO rates and the LFO depths determine how far the LFOs will sweep across their delay lines from the shortest delays to the longest delays (the LFO excursions). The Tap Delays specify the average amount of delay of the LFO modulated delay lines, or in other words the delay to the center of the LFO excursion. The center of LFO excursion can not move smoothly. Changing the center of LFO excursion creates discontinuities in the tapped signal. It is therefore a good idea to adjust the Tap Dly parameter to a reasonable setting (one which gives enough delay for the maximum LFO excursion), then leave it. Modulating Tap Dly will produce unwanted zipper noise. If you increase the LFO modulation depth or reduce the LFO rate to a point where the LFO excursion exceeds the specified Tap Dly, the center of LFO excursion will be moved up, and again cause signal discontinuities. However, if enough Tap Dly is specified, Depth and Rate will be modulated smoothly.

As the LFOs sweep across the delay lines, the signal will change pitch. The pitch will change with a triangular envelope (rise-fall-rise-fall) or with a trapezoidal envelope (rise-hold-fall-hold). You can choose the pitch envelope with the Pitch Env parameter. Unfortunately rate and depth cannot be smoothly modulated when set to the “Trapzoid” setting.

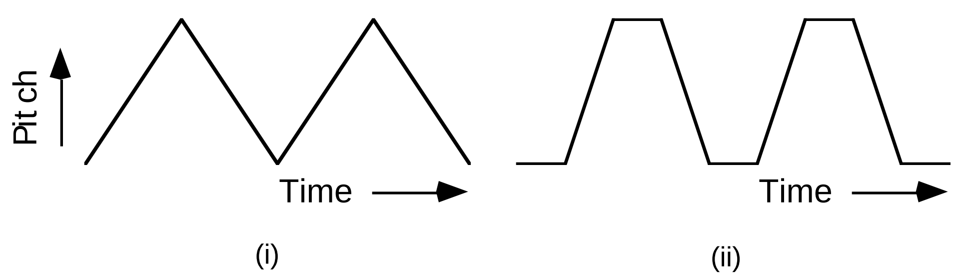


Figure 10-16 Pitch Envelopes (i) Triangle and (ii) Trapzoid

Parameters for Chorus 1

Page 1

Wet/Dry	-100 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	-100 to 100%		
Xcouple	0 to 100%		
HF Damping	16 Hz to 25088 Hz	Pitch Env	Triangle or Trapzoid

Page 2

Tap Lvl	-100 to 100%	LFO Rate	0.01 to 10.00 Hz
Tap Dly	0.0 to 1000.0 ms	LFO Depth	0.0 to 50.0 ct
		L/R Phase	0.0 to 360.0 deg

Parameters for Chorus 2

Page 1

Wet/Dry	-100 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	-100 to 100%		
Xcouple	0 to 100%		
HF Damping	16 Hz to 25088 Hz	Pitch Env	Triangle or Trapzoid

Page 2

Tap1 Lvl	-100 to 100 %	Tap1 Dly	4.0 to 1000.0 ms
Tap2 Lvl	-100 to 100 %	Tap2 Dly	4.0 to 1000.0 ms
Tap3 Lvl	-100 to 100 %	Tap3 Dly	4.0 to 1000.0 ms

Page 3

LFO1 Rate	0.01 to 10.00 Hz	LFO1 LRPhs	0.0 to 360.0 deg
LFO2 Rate	0.01 to 10.00 Hz	LFO2 LRPhs	0.0 to 360.0 deg
LFO3 Rate	0.01 to 10.00 Hz	LFO3 LRPhs	0.0 to 360.0 deg
LFO1 Dpth	0.0 to 50.0 ct		
LFO2 Dpth	0.0 to 50.0 ct		
LFO3 Dpth	0.0 to 50.0 ct		

Parameters for Dual Chorus 1

Page 1

L Wet/Dry	-100 to 100%wet	R Wet/Dry	-100 to 100%wet
L Out Gain	Off, -79.0 to 24.0 dB	R Out Gain	Off, -79.0 to 24.0 dB
L Fdbk Lvl	-100 to 100%	R Fdbk Lvl	-100 to 100%
Xcouple	0 to 100%		

Page 2

L Tap Lvl	-100 to 100%	R Tap Lvl	-100 to 100%
L Tap Pan	-100 to 100%	R Tap Pan	-100 to 100%
L LFO Rate	0.01 to 10.00 Hz	R LFO Rate	0.01 to 10.00 Hz
L LFODepth	0.0 to 50.0 ct	R LFO Depth	0.0 to 50.0 ct
L Tap Dly	0.0 to 1000.0 ms	R Tap Dly	0.0 to 1000.0 ms
L HF Damp	16 Hz to 25088 Hz	R HF Damp	16 Hz to 25088 Hz

Page 3

L PitchEnv	Triangle or Trapzoid	R PitchEnv	Triangle or Trapzoid

Parameters for Dual Chorus 2

Page 1

L Wet/Dry	-100 to 100%wet	R Wet/Dry	-100 to 100%wet
L Out Gain	Off, -79.0 to 24.0 dB	R Out Gain	Off, -79.0 to 24.0 dB
L Fdbk Lvl	-100 to 100%	R Fdbk Lvl	-100 to 100%
Xcouple	0 to 100%		

Page 2

L Tap1 Lvl	-100 to 100 %	R Tap1 Lvl	-100 to 100 %
L Tap2 Lvl	-100 to 100 %	R Tap2 Lvl	-100 to 100 %
L Tap3 Lvl	-100 to 100 %	R Tap3 Lvl	-100 to 100 %
L Tap1 Pan	-100 to 100 %	R Tap1 Pan	-100 to 100 %
L Tap2 Pan	-100 to 100 %	R Tap2 Pan	-100 to 100 %
L Tap3 Pan	-100 to 100 %	R Tap3 Pan	-100 to 100 %

Page 3

L LFO1Rate	0.01 to 10.00 Hz	R LFO1Rate	0.01 to 10.00 Hz
L LFO2Rate	0.01 to 10.00 Hz	R LFO2Rate	0.01 to 10.00 Hz
L LFO3Rate	0.01 to 10.00 Hz	R LFO3Rate	0.01 to 10.00 Hz
L LFO1Dpth	0.0 to 50.0 ct	R LFO1Dpth	0.0 to 50.0 ct
L LFO2Dpth	0.0 to 50.0 ct	R LFO2Dpth	0.0 to 50.0 ct
L LFO3Dpth	0.0 to 50.0 ct	R LFO3Dpth	0.0 to 50.0 ct

Page 4

L Tap1 Dly	0.0 to 1000.0 ms	R Tap1 Dly	0.0 to 1000.0 ms
L Tap2 Dly	0.0 to 1000.0 ms	R Tap2 Dly	0.0 to 1000.0 ms
L Tap3 Dly	0.0 to 1000.0 ms	R Tap3 Dly	0.0 to 1000.0 ms
L HF Damp	16 Hz to 25088 Hz	R HF Damp	16 Hz to 25088 Hz
L PitchEnv	Triangle or Trapzoid	R PitchEnv	Triangle or Trapzoid

Wet/Dry The relative amount of input (dry) signal and chorus (wet) signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input. When set to 100%, the output is all wet. Negative values polarity invert the wet signal.

Out Gain The overall gain or amplitude at the output of the effect.

Fdbk Level The level of the feedback signal into the delay line. The feedback signal is taken from the LFO1 delay tap. Negative values polarity invert the feedback signal.

Xcouple Controls how much of the left channel input and feedback signals are sent to the right channel delay line and vice versa. At 50%, equal amounts from both channels are sent to both delay lines. At 100%, the left feeds the right delay and vice versa.

HF Damping The amount of high frequency content of the signal that is sent into the delay lines. This control determines the cutoff frequency of the one-pole (-6dB/octave) lowpass filter.

Pitch Env The pitch of the chorus modulation can be made to follow a triangular "Triangle" envelope (rise-fall-rise-fall) or a trapezoidal "Trapzoid" envelope (rise-hold-fall-hold).

Tap Lvl Levels of the LFO modulated delay taps. Negative values polarity invert the signal. Setting any tap level to 0% effectively turns off the delay tap. Since these controls allow the full input level to pass through all the delay taps, a 100% setting on all the summed taps will significantly boost the wet signal relative to dry. A 50% setting may be more reasonable.

Tap Pan The left or right output panning of the delay taps. The range is -100% for fully left to 100% for fully right. Setting the pan to 0% sends equal amounts to both left and right channels for center or mono panning. [Dual Chorus 1 & 2 only]

LFO Rate Used to set the speeds of modulation of the delay lines. Low rates increase LFO excursion (see LFO Dpth below). If Pitch Env is set to "Trapzoid", you will be unable to put the rate on an FXMod or otherwise change the rate without introducing discontinuities (glitches or zippering) to your output signal. The triangular "Triangle" Pitch Env setting does allow smooth rate modulation, provided you've specified enough delay.

LFO Depth The maximum depths of detuning of the LFO modulated delay lines. The depth controls range from 0 to 50 cents. (There are 100 cents in a semitone.) If you do not have enough delay specified with Tap Dly to get the depth you've dialed up, then Tap Dly will be forced to increase (with signal disconinuities if signal is present). The LFOs move a tap back and forth across the delay lines to shift the pitch of the tapped signal. The maximum distance the taps get moved from the center position of the LFO is called the LFO excursion. Excursion is calculated from both the LFO depth and rate settings. Large depths and low rates produce large excursions. If Pitch Env is set to "Trapzoid", you will be unable to put the depth on an FXMod or otherwise change the depth without introducing discontinuities (glitches or zippering) to your output signal. The triangular "Triangle" Pitch Env setting does allow smooth depth modulation, provided you've specified enough delay.

Tap Dly The average delay length, or the delay to the center of the LFO sweep. If the delay is shorter than the LFO excursion, then the Tap Dly will be forced to a longer length equal to the amount of required excursion (the parameter display will not change though). Changing this parameter while signal is present will cause signal discontinuities. It's best to set and forget this one. Set it long enough so that there are no discontinuities with the largest Depth and lowest Rates that you will be using.

L/R Phase (Or **LFOn LRPhs**) In the stereo Chorus 1 and Chorus 2, the relative phases of the LFOs for the left and right channels may be adjusted.

154 Flanger 1
155 Flanger 2

Multi-tap flangers

PAUs: 1 for Flanger 1
2 for Flanger 2

Flanger 1 is a 1 processing allocation unit (PAU) multi-sweep Thru-zero flanger effect with two LFOs per channel.

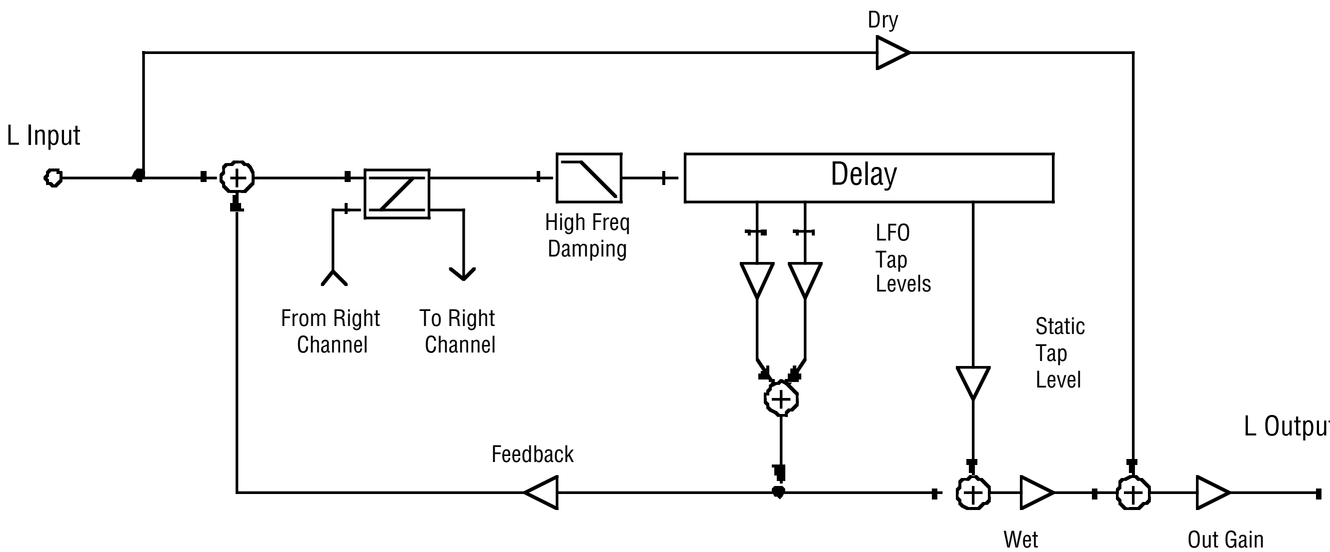


Figure 10-17 Simplified block diagram of the left channel of Flanger 1 (right channel is similar)

Flanger 2 is a 2 processing allocation unit (PAU) multi-sweep Thru-zero flanger effect with two LFOs per channel.

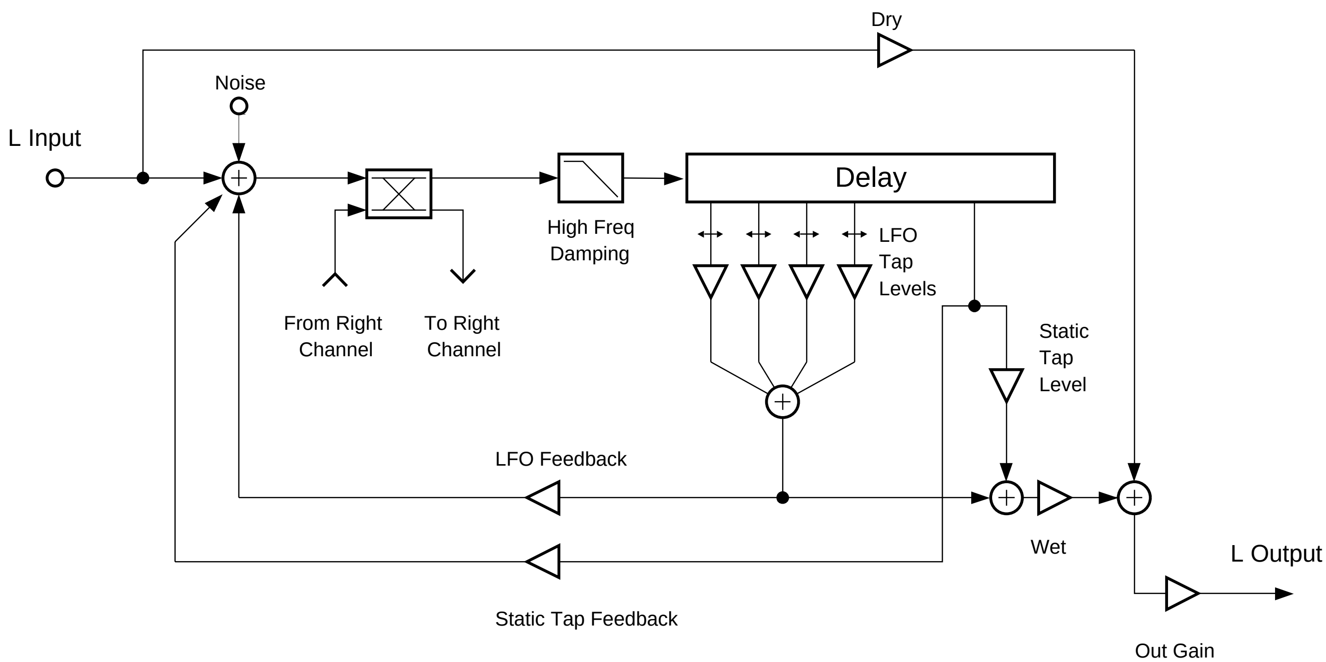


Figure 10-18 Simplified block diagram of the left channel of Flanger 2 (right channel is similar)

Flanging was originally created by summing the outputs of two un-locked tape machines while varying their sync by pressing a hand to the outside edge of one reel, thus the historic name reel-flanging. The key to achieving the flanging effect is the summing of a signal with a time-displaced replica of itself.

Adding or subtracting a signal with a time-displaced replica of itself results in a series of notches in the frequency spectrum. These notches are equally spaced in (linear) frequency at multiples whose wavelengths are equal to the time delay. The result is generally referred to as a comb filter (the name arising from the resemblance of the spectrum to a comb). See Figure 10-18. If the levels of the signals being added or subtracted are the same, the notches will be of infinite depth (in dB) and the peaks will be up 6 dB. Flanging is achieved by time-varying the delay length, thus changing the frequencies of the notches. The shorter the delay time, the greater the notch separation. This delay time variation imparts a sense of motion to the sound. Typically the delay times are on the order of 0-5 ms. Longer times begin to get into the realm of chorusing, where the ear begins to perceive the audio output as nearly two distinct signals, but with a variable time displacement.

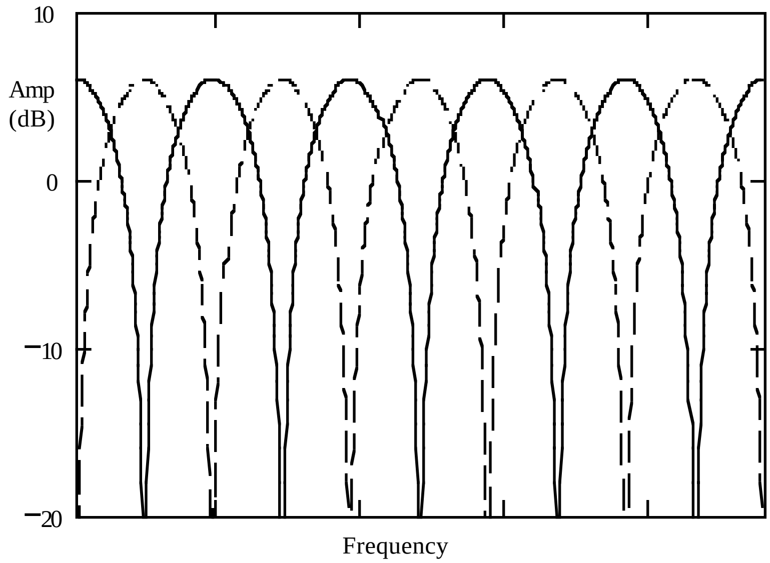


Figure 10-19 Comb Filters : Solid Line for Addition; Dashed Line for Subtraction

The heart of the flanger implemented here is a multi-tap delay line. You can set the level of each tap as a percentage of the input level, and the level may be negative (phase inverting). One tap is a simple static delay over which you can control the length of delay (from the input tap). Four of the taps can have their lengths modulated up and down by a low frequency oscillator (LFO). You are given control of the rate of the LFOs, how far each LFO can sweep through the delay line, and the relative phases of the LFOs. (i.e. Where is the LFO in its sweep: going away from the input tap or coming toward it?)

The flanger uses tempo units (based on the sequencer tempo or MIDI clock if you like), together with the number of tempo beats per LFO cycle. Thus if the tempo is 120 bpm (beats per minute) and the LFO Period is set to 1, the LFOs will pass through 120 complete cycles in a minute or 2 cycles per second (2 Hz). Increasing the LFO Period increases the period of the LFOs (slows them down). An LFO Period setting of 16 will take 4 measures (in 4/4 time) for a complete LFO oscillation.

You can set how far each LFO can sweep through the delay line with the excursion controls (Xcurs). The excursion is the maximum distance an LFO will move from the center of its sweep, and the total range of an LFO is twice the excursion. You set the delay to the center of LFO excursion with the Dly parameters. The excursion and delay controls both have coarse and fine adjustments. By setting the excursion to zero length, the LFO delay tap becomes a simple static tap with its length set to the minimum tap length. Note that modifying the delay to the center of LFO excursion will result in a sudden change of delay length and consequently, a discontinuity in the signal being read from the delay line. This can produce a characteristic zippering effect. The Dly parameters should be as long as the Xcurs parameters or longer, or else changing (or modulating) the excursion will force the center of LFO excursion to move with the resulting signal discontinuities. The static delay tap does not suffer the zippering problem, and changes to its length will occur smoothly. You can assign the static delay tap to a continuous controller and use the controller to do manual flanging. Figure 4 shows the delay line for a single LFO.

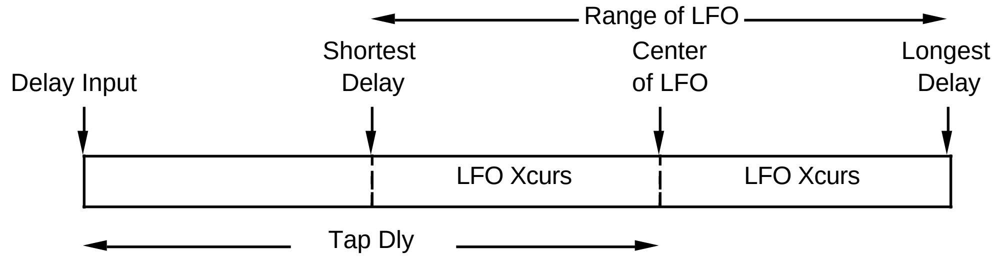


Figure 10-20 Delay for a Single LFO

Consider a simple example where you have an LFO tap signal being subtracted from the static delay tap signal. If the delays are set such that at certain times both taps are the same length, then both taps have the same signal and the subtraction produces a null or zero output. The effect is most pronounced when the static tap is set at one of the ends of the LFO excursion where the LFO tap motion is the slowest. This is the classic Thru-Zero flanger effect. Adding other LFO taps to the mix increases the complexity of the final sound, and obtaining a true Thru-Zero effect may take some careful setting of delays and LFO phases. The flanger has a Wet/Dry control as well, which can further add complexity to the output as the dry signal is added to various delayed wet components for more comb filtering.

When using more than one LFO, you can set up the phase relationships between each of the LFOs. The LFOs of the left channel and the LFOs of the right channel will be set up in the same phase relationship except that you may offset the phases of the right channel as a group relative to the left channel (L/R Phase). L/R Phase is the only control which treats left and right channels differently and has a significant effect on the stereo image. If you have tempo set to the system tempo, the phases will maintain their synchronization with the tempo clock. At the beat of the tempo clock, a phase set to 0° will be at the center of the LFO excursion and moving away from the delay input.

Regenerative feedback has been incorporated in order to produce a more intense resonant effect. The signal which is fed back is from the first LFO delay tap (LFO1), but with its own level control (Fdbk Level). In-phase spectral components arriving at the summer add together, introducing a series of resonant peaks in the frequency spectrum between the notches. The amplitude of these peaks depends on the degree of feedback and can be made very resonant.

Cross-coupling (Xcouple) allows the signals of the right and left channels to be mixed or swapped. The cross-coupling is placed after the summation of the feedback to the input signal. When feedback and cross-coupling are turned up, you will get a ping-pong effect between right and left channels.

A lowpass filter (HF Damping) right before the input to the delay line is effective in emulating the classic sounds of older analog flangers with their limited bandwidths (typically 5-6kHz).

As stated previously, it is the movement of the notches created in the frequency spectrum that give the flanger its unique sound. It should be obvious that sounds with a richer harmonic structure will be effected in a much more dramatic way than harmonically starved sounds. Having more notches, i.e. a greater ‘notch-density’, should produce an even more intense effect. This increase in notch-density may be achieved by having a number of modulating delay lines, all set at the same rate, but different depths. Setting the depths in a proportianally related way results in a more pleasing effect.

An often characteristic effect of flanging is the sound of system noise being flanged. Various pieces of analog gear add noise to the signal, and when this noise passes through a flanger, you can hear the noise “whooshing.” In the K2600, the noise level is very low, and in fact if no sound is being played, there is no noise at all at this point in the signal chain. To recreate the effect of system noise flanging, white noise may be added to the input of the flanger signal (Flanger 2 only). White noise has a lot of high frequency content and may sound too bright. The noise may be tamed with a first order lowpass filter.

Parameters for Flanger 1

Page 1

Wet/Dry	-100 to 100% wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	-100 to 100%	LFO Tempo	System, 1 to 255 BPM
Xcouple	0 to 100%	LFO Period	1/24 to 32 bts
HF Damping	16 to 25088 Hz		

Page 2

StatDlyLvl	-100 to 100%	L/R Phase	0.0 to 360.0 deg
LFO1 Level	-100 to 100%	LFO1 Phase	0.0 to 360.0 deg
LFO2 Level	-100 to 100%	LFO2 Phase	0.0 to 360.0 deg

Page 3

StatDlyCrs	0.0 to 228.0 ms		
StatDlyFin	-127 to 127 samp		
Xcurs1 Crs	0.0 to 228.0 ms	Dly1 Crs	0.0 to 228.0 ms
Xcurs1 Fin	-127 to 127 samp	Dly1 Fin	-127 to 127 samp
Xcurs2 Crs	0.0 to 228.0 ms	Dly2 Crs	0.0 to 228.0 ms
Xcurs2 Fin	-127 to 127 samp	Dly2 Fin	-127 to 127 samp

Parameters for Flanger 2

Page 1

Wet/Dry	-100 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
LFO Fdbk	-100 to 100%	Stat Fdbk	-100 to 100%
Xcouple	0 to 100%	LFO Tempo	System, 1 to 255 BPM
HF Damping	16 Hz to 25088 Hz	LFO Period	1/24 to 32 bts

Page 2

Noise Gain	Off, -79.0 to -30.0 dB	Noise LP	16 to 25088 Hz
StatDlyLvl	-100 to 100 %	L/R Phase	0.0 to 360.0 deg
LFO1 Level	-100 to 100 %	LFO1 Phase	0.0 to 360.0 deg
LFO2 Level	-100 to 100 %	LFO2 Phase	0.0 to 360.0 deg
LFO3 Level	-100 to 100 %	LFO3 Phase	0.0 to 360.0 deg
LFO4 Level	-100 to 100 %	LFO4 Phase	0.0 to 360.0 deg

Page 3

StatDlyCrs	0.0 to 228.0 ms		
StatDlyFin	-127 to 127 samp		
Xcurs1 Crs	0.0 to 228.0 ms	Xcurs3 Crs	0.0 to 228.0 ms
Xcurs1 Fin	-127 to 127 samp	Xcurs3 Fin	-127 to 127 samp
Xcurs2 Crs	0.0 to 228.0 ms	Xcurs4 Crs	0.0 to 228.0 ms
Xcurs2 Fin	-127 to 127 samp	Xcurs4 Fin	-127 to 127 samp

Page 4

Dly1 Crs	0.0 to 228.0 ms	Dly3 Crs	0.0 to 228.0 ms
Dly1 Fin	-127 to 127 samp	Dly3 Fin	-127 to 127 samp
Dly2 Crs	0.0 to 228.0 ms	Dly4 Crs	0.0 to 228.0 ms
Dly2 Fin	-127 to 127 samp	Dly4 Fin	-127 to 127 samp

Wet/Dry The relative amount of input signal and flanger signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet. Negative values polarity invert the wet signal.

Out Gain The overall gain or amplitude at the output of the effect.

Fdbk Level The level of the feedback signal into the delay line. The feedback signal is taken from the LFO1 delay tap. Negative values polarity invert the feedback signal.

Xcouple How much of the left channel input and feedback signals are sent to the right channel delay line and vice versa. At 50%, equal amounts from both channels are sent to both delay lines. At 100%, the left feeds the right delay and vice versa. Xcouple has no effect if Fdbk Level is set to 0%.

HF Damping The amount of high frequency content of the signal sent into the delay lines. This control determines the cutoff frequency of the one-pole (-6dB/octave) lowpass filters.

LFO Tempo Basis for the rates of the LFOs, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LFO Period Sets the LFO rate based on the Tempo determined above: the number of beats corresponding to one period of the LFO cycle. For example, if the LFO Period is set to “4”, the LFOs will take four beats to pass through one oscillation, so the LFO rate will be 1/4th of the Tempo setting. If it is set to “6/24” (=1/4), the LFO will oscillate four times as fast as the Tempo. At “0”, the LFOs stop oscillating and their phase is undetermined (wherever they stopped).

Noise Gain The amount of noise (dB relative to full scale) to add to the input signal. In many flangers, you can hear the noise floor of the signal being flanged, but in the K2600, if there is no input signal, there is no noise floor unless it is explicitly added. [Flanger 2 only]

Noise LP The cut-off frequency of a one pole lowpass filteracting on the noise injection signal. The lowpass removes high frequencies from an otherwise pure white noise signal. [Flanger 2 only]

StatDlyCrs The nominal length of the static delay tap from the delay input. The name suggests the tap is stationary, but it can be connected to a control source such as a data slider, a ribbon, or a

VAST function to smoothly vary the delay length. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective.

StatDlyFin A fine adjustment to the static delay tap length. The resolution is one sample.

StatDlyLvl The level of the static delay tap. Negative values polarity invert the signal. Setting any tap level to 0% turns off the delay tap.

Xcurs *n* Crs The LFO excursion controls set how far the LFO modulated delay taps can move from the center of their ranges. The total range of the LFO sweep is twice the excursion. If the excursion is set to 0, the LFO does not move and the tap behaves like a simple delay line set to the minimum delay. The excursion cannot be made longer than than the delay to the center of excursion (see Dly Crs & Dly Fin below) because delays cannot be made shorter than 0. If you attempt longer excursions, the length of the Dly Crs/Fin will be forced to increase (though you will not see the increased length displayed in the Dly Crs/Fin parameters). The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. This parameter is a coarse adjustment for the excursion.

Xcurs *n* Fin A fine adjustment for the LFO excursions. The resolution is one sample.

Dly *n* Crs The delay to the center of LFO tap range. The maximum delay will be this delay plus the LFO excursion delay. The minimum delay will be this delay minus the LFO excursion delay. Since delays cannot be less than 0 ms in length, the this delay length will be increased if LFO excursion is larger than this delay length. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. This parameter is a coarse adjustment for the delay.

Dly *n* Fin A fine adjustment to the minimum delay tap lengths. The resolution is one sample.

LFO*n* Level The levels of the LFO modulated delay taps. Negative values polarity invert the signal. Setting any tap level to 0% turns off the delay tap.

LFO*n* Phase The phase angles of the LFOs relative to each other and to the system tempo clock, if turned on (see Tempo). For example, if one LFO is set to 0° and another is set to 180°, then when one LFO delay tap is at its shortest, the other will be at its longest. If the system tempo clock is on, the LFOs are synchronized to the clock with absolute phase. A phase of 0° will put an LFO tap at the center of its range and its lengthening.

L/R Phase Adds the specified phase angle to the right channel LFOs. In all other respects the right and left channels are symmetric. By moving this control away from 0°, the stereo sound field is broken up and a stereo image becomes difficult to spatially locate. The effect is usually described as “phasey.” It tends to impart a greater sense of motion.

Algorithms 156–160: Phasers

156 LFO Phaser
157 LFO Phaser Twin
158 Manual Phaser
159 Vibrato Phaser
160 SingleLFO Phaser

A variety of single notch/bandpass Phasers

PAUs: 1 each

A simple phaser is an algorithm which produces an vague swishing or phasey effect. When the phaser signal is combined with the dry input signal or the phaser is fed back on itself, peaks and/or notches can be produced in the filter response making the effect much more pronounced. Most of the phaser algorithms presented here have built in low frequency oscillators (LFOs) to generate the motion of the phasers. In the case of Manual Phaser, the phaser motion is left to you.

A phaser uses a special filter called an allpass filter to modify the phase response of a signal's spectrum without changing the amplitude of the spectrum. Okay, that was a bit of a mouthful — so what does it mean? As the term "allpass filter" suggests, the filter by itself does not change the amplitude response of a signal passing through it. An allpass filter does not cut or boost any frequencies. An allpass filter does cause some frequencies to be delayed a little in time, and this small time shift is also known as a phase change. The frequency where the phase change has its greatest effect is a parameter that you can control. By modulating the frequency of the phaser, you get the swishy phaser sound. With a modulation rate of around 6 Hz, an effect similar to vibrato may be obtained, but only in a limited range of filter frequencies.

By adding the phaser output to the dry input using, for example, a Wet/Dry parameter, you can produced peaks and notches in the frequency response. At frequencies where the phaser is "in phase" with the dry signal, the signal level doubles (or there is a 6 dB level increase approximately). At frequencies where the phaser and dry signals are "out of phase", the two signals cancel each other out and there is a notch in the frequency response. You can get a complete notch when Wet/Dry is set to 50%. If subtraction is used instead of addition by setting Wet/Dry to -50%, then the notches become peaks and the peaks become notches.

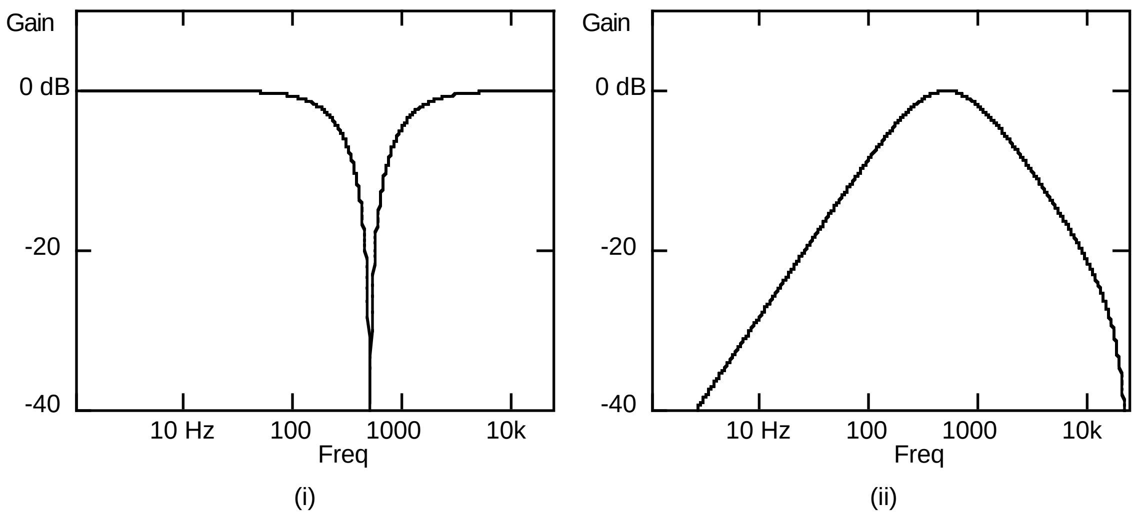


Figure 10-21 Response of typical phaser with (i) Wet/Dry = 50% and (ii) WetDry = -50%.

Some of the phaser algorithms have feedback. When feedback is used, it can greatly exaggerate the peaks and notches, producing a much more resonant sound.

LFO Phasor is a simple phaser algorithm with Wet/Dry and Fdbk Level parameters. Two LFOs are built in to control the filter frequency and the depth of the resulting notch. You can control the depths, rates, and phases of both the LFOs. The algorithm is stereo so the relative phases of the LFOs for the left and right channels can be set. When setting the LFO which controls the filter frequency, you specifiy the center frequency around which the LFO will modulate and the depth of the LFO. The depth specifies how many cents (hundredths of a semitone) to move the filter frequency up and down. The NotchDepth parameter provides an alternative way of combining wet and dry phaser signals to produce a notch. In this case the parameter specifies the depth of the notch in decibels (dB). The depth of the notch can be modulated with the notch LFO. The notch LFO is completely independent of the frequency LFO. The rates of the LFOs may be different. The relative phases of the notch and frequency LFOs (N/F Phase) only has meaning when the LFOs are running at the same rate. As with all KDFX LFO phases, it is not a recommended to directly modulate the phase settings with an FXMod.

SingleLFO Phaser is identical to LFO Phaser except that the notch and frequency LFOs always run at the same rate.

As mentioned earlier, Manual Phaser leaves the phaser motion up to you, so it has no built in LFOs. Manual Phaser has a Notch/BP parameter which produces a complete notch at the center frequency when Wet/Dry is set to -100% and a resonant bandpass when set to 100%. At 0% the signal is dry. To get phaser motion, you have to change the filter center frequencies (left and right channels) yourself. The best way to do this is with an FXMod. There are also feedback parameters for the left and right channels.

LFO Phaser Twin produces a pair of notches separated by a spectral peak. The center frequency parameter sets the frequency of the center peak. Like LFO Phaser, the filter frequency can be modulated with a built in LFO. The Notch/Dry parameter produces a pair of notches when set to 100%. The output signal is dry when set to 0% and at 200%, the signal is a pure (wet) allpass response. LFO Phaser Twin does not have Out Gain or feedback parameters.

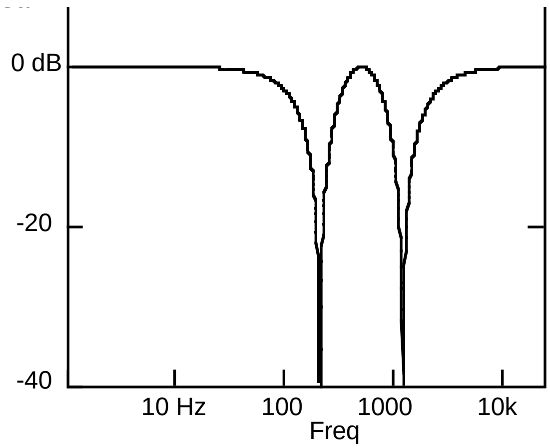


Figure 10-22 Response of LFO Phaser Twin with Wet/Dry set to 100%.

The Vibrato Phaser algorithm has a couple of interesting twists. The bandwidth of the phaser filter can be adjusted exactly like a parametric EQ filter. The built in LFO can be made to run at audio rates by multiplying the LFO Rate parameter with the Rate Scale parameter. Running the LFO at audio rates produces strange frequency modulation effects. The In Width controls how the stereo input signal is routed through the effect. At 100% In Width, left input is processed to the left output, and right to right. Lower In Width values narrow the input stereo field until at 0%, the processing is mono. Negative values reverse left and right channels. The dry signal is not affected by In Width. As described earlier setting Wet/Dry to 50% will produce a full notch. At -50% Wet/Dry, you get a bandpass.

Parameters for LFO Phaser

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	-100 to 100%		

Page 2

CenterFreq	16 to 25088 Hz	NotchDepth	-79.0 to 6.0 dB
FLFO Depth	0 to 5400 ct	NLFO Depth	0 to 100 %
FLFO Rate	0.00 to 10.00 Hz	NLFO Rate	0.00 to 10.00 Hz
FLFO LRPhs	0.0 to 360.0 deg	NLFO LRPhs	0.0 to 360.0 deg
		N/F Phase	0.0 to 360.0 deg

Parameters for SingleLFO Phaser

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	-100 to 100%		

Page 2

LFO Rate	0.00 to 10.00 Hz	N/F Phase	
CenterFreq	16 to 25088 Hz	NotchDepth	-79.0 to 6.0 dB
FLFO Depth	0 to 5400 ct	NLFO Depth	0 to 100 %
FLFO LRPhs	0.0 to 360.0 deg	NLFO LRPhs	0.0 to 360.0 deg

Wet/Dry The amount of phaser (wet) signal relative to unaffected (dry) signal as a percent.

Out Gain The output gain in decibels (dB) to be applied to the combined wet and dry signals.

Fdbk Level The phaser output can be added back to its input to increase the phaser resonance. Negative values polarity invert the feedback signal.

LFO Rate The rate of both the center frequency LFO and the notch depth LFO for the SingleLFO Phaser algorithm.

CenterFreq The nominal center frequency of the phaser filter. The frequency LFO modulates the phaser filter centered at this frequency.

FLFO Depth The depth in cents that the frequency LFO sweeps the phaser filter above and below the center frequency.

FLFO Rate The rate of the center frequency LFO for the LFO Phaser algorithm.

FLFO LRPhs Sets the phase difference between the left and right channels of the center frequency LFO. A setting of 180 degrees results in one being at a at the minimum frequency while the other channel is at the maximum.

NotchDepth The nominal depth of the notch. The notch depth LFO modulates the depth of the notch. For maximum LFO depth, set NotchDepth to 0 dB and NLFO Depth to 100%.

NLFO Depth The excursion of the notch depth LFO in units of percentage of the total range. The depth of the LFO is limited to the range of the NotchDepth parameter such that a full 100% modulation is only possible with the NotchDepth is at the center of its range (0 dB).

NLFO Rate The rate of the notch depth LFO for the LFO Phaser algorithm.

NLFO LRPhs The phase difference between the left and right channels of the notch depth LFO. A setting of 180 degrees results in one channel being at highest amplitude while the other channel is at lowest amplitude.

N/F Phase The phase difference between the notch depth and center frequency LFOs. For LFO Phaser, this parameter is largely meaningless unless the FMod Rate and NMod Rate are set identically.

Parameters for Manual Phaser

Page 1

Notch/BP	-100 to 100%	Out Gain	Off, -79.0 to 24.0 dB
L Feedback	-100 to 100%	R Feedback	-100 to 100%
L Ctr Freq	16 to 25088 Hz	R Ctr Freq	16 to 25088 Hz

Notch/BP The amount of notch depth or bandpass. At -100% there is a complete notch at the center frequency. At 100% the filter response is a peak at the center frequency. 0% is the dry unaffected signal.

Out Gain The output gain in decibels (dB) to be applied to the final output.

Feedback The phaser output can be added back to its input to increase the phaser resonance (left and right). Negative values polarity invert the feedback signal.

Ctr Freq The nominal center frequency of the phaser filter (left and right). For a true phaser effect you may want to modulate these parameters by setting up FX Mods.

Parameters for LFO Phaser Twin

Page 1

Notch/Dry	0 to 200%		
CenterFreq	16 to 25088 Hz	LFO Rate	0.00 to 10.00 Hz
LFO Depth	0 to 5400 ct	L/R Phase	0.0 to 360.0 deg

Notch/Dry The amount of phaser (wet) signal relative to unaffected (dry) signal as a percent. At 100% the phaser produces a pair of full notches above and below the center frequency. At 200% the output is a pure allpass response (no amplitude changes, but phase changes centered about the center frequency).

CenterFreq The nominal center frequency of the phaser filter. When configured for a maximum notch (Notch/Dry is 100%), the CenterFreq specifies the frequency of the peak between two notches. The LFO modulates the phaser filter centered at this frequency.

LFO Rate The rate of the phaser frequency modulating LFO in Hertz.

LFO Depth The depth in cents that the frequency LFO sweeps the phaser filter above and below the center frequency.

L/R Phase The phase difference between the left and right channels of the LFO. A setting of 180 degrees results in one being at the minimum frequency while the other channel is at the maximum.

Parameters for Vibrato Phaser

Page 1

Wet/Dry	-100 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB

Page 2

CenterFreq	16 to 25088 Hz	Bandwidth	0.010 to 5.000 oct
LFO Depth	0 to 100%	L/R Phase	0.0 to 360.0 deg
LFO Rate	0.00 to 10.00 Hz		
Rate Scale	1 to 25088x	In Width	-100 to 100%

Wet/Dry The amount of phaser (wet) signal relative to unaffected (dry) signal as a percent. When set to 50% you get a complete notch. When set to -50%, the response is a bandpass filter. 100% is a pure allpass filter (no amplitude changes, but a strong phase response).

Out Gain The output gain in decibels (dB) to be applied to the combined wet and dry signals.

CenterFreq The nominal center frequency of the phaser filter. The frequency LFO modulates the phaser filter centered at this frequency.

Bandwidth If the phaser is set to behave as a sweeping notch or bandpass, the bandwidth of the notch or bandpass is set with Bandwidth. This parameter works the same as for parametric EQ filter bandwidths.

LFO Depth The depth that the frequency LFO sweeps the phaser filter above and below the center frequency as a percent.

LFO Rate The rate of the LFO in Hertz. The LFO Rate may be scaled up by the Rate Scale parameter.

Rate Scale A rate multiplier value which may be used to increase the LFO frequency to audio rates. For example, if LFO Rate is set to 1.00 Hz and Rate Scale is set to 1047x, then the LFO frequency is 1047 x 1.00 Hz = 1047 Hz.

L/R Phase Sets the phase difference between the left and right channels of the center frequency LFO. A setting of 180 degrees results in one being at a at the minimum frequency while the other channel is at the maximum.

In Width The width of the stereo field that passes through the stereo phaser filtering. This parameter does not affect the dry signal. When set to 100%, the left and right channels are processed to their respective outputs. Smaller values narrow the stereo image until at 0% the input channels are summed to mono and set to left and right outputs. Negative values interchange the left and right channels.

Combination Algorithms

700 Chorus+Delay
701 Chorus+4Tap
703 Chor+Dly+Reverb
706 Flange+Delay
707 Flange+4Tap
709 Flan+Dly+Reverb
722 Pitcher+Chor+Dly
723 Pitcher+Flan+Dly

A family of combination effect algorithms ("+")

PAUs: 1 or 2

Signal Routing (2 effects)

The algorithms listed above with 2 effects can be arranged in series or parallel. Effect A and B are respectively designated as the first and second listed effects in the algorithm name. The output effect A is wired to the input of effect B, and the input into effect B is a mix of effect A and the algorithm input dry signal. The effect B input mix is controlled by a parameter A/Dry>B. where A is effect A, and B is effect B. For example, in Chorus+Delay, the parameter name is "Ch/Dry>Dly". The value functions much like a wet/dry mix where 0% means that only the algorithm input dry signal is fed into effect B (putting the effects in parallel), and 100% means only the output of effect A is fed into effect B (putting the effects in series). See Figure 10-23 for signal flow of Chorus+4Tap as an example.

Both effect A and B outputs are mixed at the algorithm output to become the wet signal. These mix levels are controlled with the 2 parameters that begin with "Mix". These allow only one or both effect outputs to be heard. Negative mix amounts polarity invert the signal which can change the character of each effect when mixed together or with the dry signal. The Wet/Dry parameter adjusts the balance between the sum of both effects determined by the Mix parameters, and the input dry signal. Negative Wet/Dry values polarity invert the summed wet signal relative to dry.

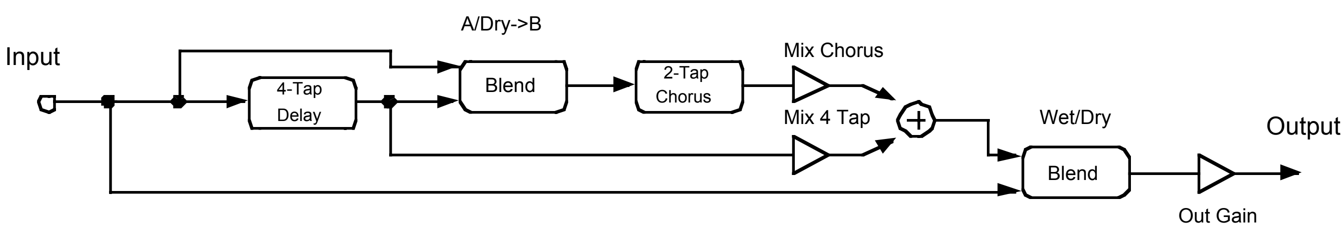


Figure 10-23 An example of routing using Chorus+4Tap

Parameters for Two-effect Routing

Page 1

Wet/Dry	-100 to 100 %	Out Gain	Off; -79.0 to 24.0 dB
Mix Effect	-100 to 100 %		
Mix Effect	-100 to 100 %		
		A/Dry->B	0 to 100%

Mix Effect Adjusts the amount of each effect that is mixed together as the algorithm wet signal. Negative values polarity invert that particular signal.

A/Dry->B This parameter controls how much of the A effect is mixed with dry and fed into the B effect. A and B are designated in the algorithm name. This control functions like a wet/dry mix, where 0% is completely dry and 100% is effect A only.

Signal Routing (3 effects)

The algorithms listed above with 3 effects allow serial or parallel routing between any two effects. Effects A, B, and C are designated respectively by their order in the algorithm name. Effect A is wired to the input of effect B and C, and effect B is wired into effect C. The input of effect B is a mix between effect A and the algorithm dry input. The input into effect C is a three-way mix between effect A, effect B, and the dry signal.

Like in the 2 effect routing, the input of effect B is controlled by a parameter A/Dry>B. where A is effect A, and B is effect B. For example, in Chor+Dly+Rvb, the parameter name is “Ch/Dry>Dly”.

The input into effect C is controlled by 2 parameters named A/B ->* and */Dry->C where A, B, and C correspond to the names of effects A, B, and C. The first parameter mixes effect A and B into a temporary buffer represented by the symbol “*”. The second parameter mixes this temporary buffer “*” with the dry signal to be fed into effect C. These mixing controls function similarly to Wet/Dry parameters. A setting of 0% only mixes the denominator, while 100% only mixes the numerator. Negative values polarity invert the signal associated with the numerator.

Effects A, B, and C outputs are mixed at the algorithm output to become the wet signal. Separate mixing levels are provided for left and right channels, and are named “L Mix” or “R Mix”. Negative mix amounts polarity invert the signal which can change the character of each effect when mixed together or with the dry signal. The Wet/Dry parameter adjusts the balance between the sum of all effects determined by the Mix parameters, and the input dry signal. Negative Wet/Dry values polarity invert the summed wet signal relative to dry.

Parameters for Three-effect Routing

Page 1

Wet/Dry	-100 to 100 %	Out Gain	Off; -79.0 to 24.0 dB
L Mix Effect A	-100 to 100 %	R Mix Effect A	-100 to 100 %
L Mix Effect B	-100 to 100 %	R Mix Effect B	-100 to 100 %
L Mix Effect C	-100 to 100 %	R Mix Effect C	-100 to 100 %

Page 2

A/Dry>B	-100 to 100 %	A/Dry>B	-100 to 100 %
A/B ->*	-100 to 100 %	A/B ->*	-100 to 100 %

Mix Effect Left and Right. Adjusts the amount of each effect that is mixed together as the algorithm wet signal. Separate left and right controls are provided. Negative values polarity invert that particular signal.

A/Dry>B This parameter controls how much of the A effect is mixed with dry and fed into the B effect. A and B are designated in the algorithm name. This control functions like a wet/dry mix, where 0% is completely dry and 100% is effect A only.

A/B ->* This parameter is first of two parameters that control whet is fed into effect C. This adjusts how much of the effect A is mixed with effect B, the result of which is represented as the symbol “*”. 0% is completely B effect, and 100% is completely A effect. negative values polarity invert the A effect.

***/Dry->C** This parameter is the second of two parameters that control whet is fed into effect C. This adjusts how much of the “*” signal (sum of effects A and B determined by A/B ->*) is mixed with the dry signal and fed into effect C. 0% is completely dry signal, and 100% is completely “*” signal.

Individual Effect Components

Chorus

The choruses are basic 1 tap dual choruses. Separate LFO controls are provided for each channel. Slight variations between algorithms may exist. Some algorithms offer separate left and right feedback controls, while some offer only one for both channels. Also, cross-coupling and high frequency damping may be offered in some and not in others. Parameters associated with chorus control begin with “Ch” in the parameter name. A general description of chorus functionality can be found in the Chorus section.

Parameters for Chorus

Page 1

Ch PtchEnv	Triangle or Trapzoid		
Ch Rate L	0.01 to 10.00 Hz	Ch Rate R	0.01 to 10.00 Hz
Ch Depth L	0.0 to 100 ct	Ch Depth R	0.0 to 100 ct
Ch Delay L	0 to 1000 ms	Ch Delay R	0 to 1000 ms
Ch Fdbk	-100 to 100 %		
Ch Xcouple	0 to 100%	Ch HF Damp	16 to 25088 Hz

Ch Fdbk This controls the amount that the output of the chorus is fed back into the input.

All Other Parameters Refer to Chorus documentation.

Flange

The flangers are basic 1 tap dual flangers. Separate LFO controls are provided for each channel. Slight variations between algorithms may exist. Some algorithms offer separate left and right feedback controls, while some offer only one for both channels. Also, cross-coupling and high frequency damping may be offered in some and not in others. Parameters associated with chorus control begin with “Ch” in the parameter name. A general description of chorus functionality can be found in the Chorus section.

In addition to the LFO delay taps, some flangers may offer a static delay tap for creating through-zero flange effects. The maximum delay time for this tap is 230ms and is controlled by the Fl StatDly parameter. Its level is controlled by the Fl StatLvl parameter.

Parameters for Flange

Page 1

Fl Tempo	System; 1 to 255 BPM	Fl HF Damp	16 to 25088 Hz
Fl Rate	0.01 to 10.00 Hz		
Fl Xcurs L	0 to 230 ms	Fl Xcurs R	0 to 230 ms
Fl Delay L	0 to 230 ms	Fl Delay R	0 to 230 ms
Fl Fdbk L	-100 to 100 %	Fl Fdbk R	-100 to 100 %
Fl Phase L	0 to 360 deg	Fl Phase R	0 to 360 deg

Page 2

Fl HF Damp	16 to 25088 Hz
Fl Xcouple	0 to 100%
Fl StatDly	0 to 230 ms
Fl StatLvl	-100 to 100 %

Fl Phase Left and Right. These adjust the corresponding LFO phase relationships between themselves and the internal beat clock.

Fl StatDly Sets the delay time for the non-moving delay tap for through-zero flange effects.

Fl StatLvl Adjusts the mix amount for the static tap. Negative values polarity invert the static tap signal.

All other parameters Refer to Flange documentation. Parameters with a 1 or 2 correspond to LFO taps organized as described above.

Delay

The Delay is a basic tempo based dual channel delay with added functionality, including image shifting, and high frequency damping. Separate left and right controls are generally provided for delay time and feedback, and laser controls. Parameters associated with Laser Verb in a combination algorithm begin with Dly.

The delay length for each channel is determined by Dly Tempo, expressed in beats per minute (BPM), and the delay length (Dly Time L and Dly Time R) of each channel is expressed in beats (bts). The tempo alters both channel delay lengths together. With the tempo in beats per minute and delay lengths in beats, you can calculate the length of a delay in seconds as beats/tempo * 60 (sec/min). Since KDFX has a limited amount of delay memory available (usually 1.5 seconds for these delays), selecting slow tempos and/or long delay lengths may cause you to run out of delay memory. At this point, each delay will pin at it’s maximum possible time. Because of this, when you slow down the tempo, you may find the delays lose their sync.

Delay regeneration is controlled by Dly Fdbk. Separate left and right feedback control is generally provided, but due to resource allocation, some delays in combinations may have a single control for both channels.

Dly FBImag and Dly HFDamp are just like the HFDamp and Image parameters found in other algorithms. Not all delays in combination algorithms will have both of these parameters due to resource allocation.

Parameters for Delay

Page 1

Dly Time L	0 to 32 bts	Dly Time R	0 to 32 bts
Dly Fdbk L	-100 to 100 %	Dly Fdbk R	-100 to 100 %
Dly HFDamp	0 to 32 bts	Dly Imag	-100 to 100 %

Dly Time Left and Right. The delay lengths of each channel in beats. The duration of a beat is specified with the Tempo parameter. The delay length in seconds is calculated as beats/ tempo * 60 (sec/min).

Dly Fdbk The amount of the output of the effect that is fed back to the input.

Dly HFDamp Controls the cutoff frequency of a 1 pole (6dB/oct slope) lopass filter in the feedback path. The filter is heard when either Dly Fdbk or LsrCntour is used.

Dly FBImag Controls the amount of image shifting during each feedback regeneration, and is heard only when Dly Fdbk is used. Small positive values shift the image to the right, while small negative values shift to the left. Larger values tend to shift the image so far that the image gets scrambled, and in some cases create ambience.

Combination 4-Tap

Combination 4-Tap is a tempo based 4 tap delay with feedback used in combination algorithms. Parameters associated with the 4 tap effect start with “4T”. The control over the feedback tap and individual output taps is essentially the same as the 4-Tap Delay BPM algorithm, with the exception that the delay times will pin at the maximum delay time instead of automatically cutting their times in half.

Parameters for Combination 4-Tap

Page 1

4T Tempo	System; 1 to 255 BPM
4T LoopLen	0 to 8 bts
4T FB Lvl	-100 to 100 %

Page 2

Tap1 Delay	0 to 8 bts	Tap3 Delay	0 to 8 bts
Tap1 Level	-100 to 100 %	Tap3 Level	-100 to 100 %
Tap1 Bal	-100 to 100 %	Tap3 Bal	-100 to 100 %
Tap2 Delay	0 to 8 bts	Tap4 Delay	0 to 8 bts
Tap2 Level	-100 to 100 %	Tap4 Level	-100 to 100 %
Tap2 Bal	-100 to 100 %	Tap4 Bal	-100 to 100 %

Reverb

The reverbs offered in these combination effects is MiniVerb. Information about it can be found in the MiniVerb documentation. Parameters associated with this reverb begin with Rv.

MiniVerb

		Rv Type	Hall1
		Rv Time	0.5 to 30.0 s; Inf
Rv DiffScl	0.00 to 2.00x	Rv Density	0.00 to 4.00x
Rv SizeScl	0.00 to 4.00x	Rv HF Damp	16 to 25088 Hz
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms

Configurable Combination Algorithms

702 Chorus<>4Tap
704 Chorus<>Reverb
705 Chorus<>LasrDly
708 Flange<>4Tap
710 Flange<>Reverb
711 Flange<>LasrDly
712 Flange<>Pitcher
713 Flange<>Shaper
714 LasrDly<>Reverb
715 Shaper<>Reverb

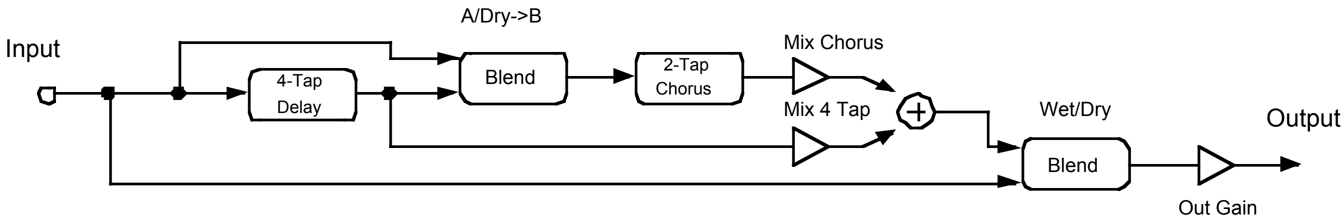
A family of combination effect algorithms

PAUs: 2

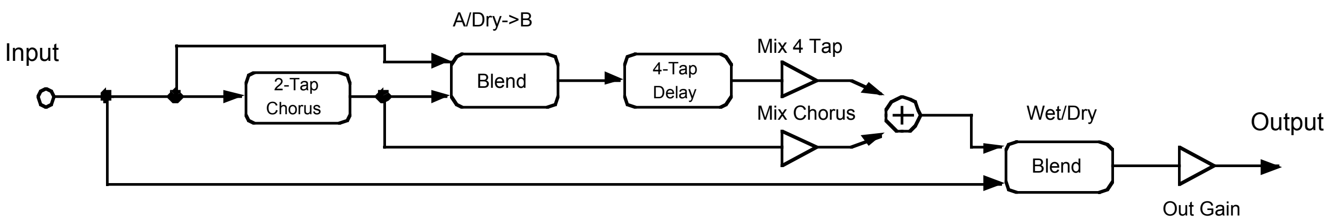
Signal Routing

Each of these combination algorithms offer 2 separate effects combined with flexible signal routing mechanism. This mechanism allows the 2 effects to either be in series bi-directionally or in parallel. This is done by first designating one effect “A”, and the other “B” where the output of effect A is always wired to effect B. A and B are assigned with the A->B cfg parameter. For example, when A->B cfg is set to Ch->Dly, then effect A is the chorus, and effect B is the delay, and the output of the chorus is wired to the input of the delay. The amount of effect A fed into effect B is controlled by the A/Dry->B parameter. This controls the balance between effect A output, and the algorithm dry input signal fed into effect B behaving much like a wet/dry mix. When set to 0%, only the dry signal is fed into B allowing parallel effect routing. At 100%, only the A output is fed into B, and at 50%, there is an equal mix of both. For an example of signal flow in the Chor<>4Tap algorithm, see Figure 10-24.

Both effect A and B outputs are mixed at the algorithm output to become the wet signal. These mix levels are controlled with the 2 parameters that begin with “Mix”. These allow only one or both effect outputs to be heard. Negative mix amounts polarity invert the signal which can change the character of each effect when mixed together or with the dry signal. The Wet/Dry parameter adjusts the balance between the sum of both effects determined by the Mix parameters, and the input dry signal. Negative Wet/Dry values polarity invert the summed wet signal relative to dry.



Configured as Ch -> 4T



Configured as 4T -> Ch

Figure 10-24 Chor<>4Tap with A->B cfg set to Ch->4T and 4T->Ch

Bi-directional Routing

Wet/Dry	-100 to 100 %	Out Gain	Off; -79.0 to 24.0 dB
Mix *Effect*	-100 to 100 %		
Mix *Effect*	-100 to 100 %		
A->B cfg	EffectA->EffectB	A/Dry->B	0 to 100%

Mix Effect Adjusts the amount of each effect is mixed together as the algorithm wet signal. Negative values polarity invert that particular signal.

A->B cfg This parameter controls the order of the effects routing. The output of effect A is wired into the input of effect B. So, when set to Ch->4T for example, effect A is chorus, and effect B is 4-tap. This is used in conjunction with the A/Dry->B parameter.

A/Dry->B This parameter controls how much of the A effect is mixed with dry and fed into the B effect. A and B are determined by the A->B cfg parameter. This works like a wet/dry mix, where 0% is completely dry and 100% is effect A only.

Individual Effect Components

Configurable Chorus and Flange

The configurable chorus and flange have 2 moving delay taps per channel. Parameters associated with chorus control begin with “Ch” in the parameter name, and those associated with flange begin with Fl. General descriptions of chorus and flange functionality can be found in the Chorus or Flange sections.

Since these effects have 2 taps per channel, control over 4 LFOs is necessary with a minimum number of user parameters (Figure 2). This is accomplished by offering 2 sets of LFO controls with three user interface modes: Dual1Tap, Link1Tap, or Link2Tap. These are selectable with the LFO cfg parameter and affect the functionality of the 2 sets of rate, depth and delay controls (and also phase and feedback controls for the flange). Each parameter is labeled with a 1 or a 2 in the parameter name to indicate to which control set it belongs. Control set 1 consists of controls whose name ends with a 1, and control set 2 consists of controls whose name ends with a 2.

In Dual1Tap mode (Figure 3), each control set independently controls 1 tap in each channel. This is useful for dual mono applications where separate control over left and right channels is desired. Control set 1 controls the left channel, and control set 2 controls the right channel. The second pair of moving delay taps are disabled in this mode. LRPhase is unpredictable unless both rates are set to the same speed. Then, the phase value is accurate only after the LFOs are reset. LFOs can be reset by either changing the LFO cfg parameter, or loading in the algorithm by selecting a preset or studio that uses it. For user-friendly LRPhase control, use either the Link1Tap or Link2Tap modes.

In Link1Tap mode (Figure 4), control set 1 controls 1 tap in both the left and right channels. Control set 2 has no affect, and the second pair of LFO delay taps are disabled. This mode is optimized for an accurate LRPhase relationship between the left and right LFOs.

In Link2Tap mode (Figure 5), control set 1 controls the first left and right pair of LFOs, while control set 2 controls the second pair. This mode uses all 4 LFOs for a richer sound, and is optimized for LRPhase relationships. Each of the 2 taps per channel are summed together at the output, and the Fdbk parameters control the sum of both LFO taps on each channel fed back to the input.

In addition to the LFO delay taps, the flange offers a static delay tap for creating through-zero flange effects. The maximum delay time for this tap is 230ms and is controlled by the Fl StatDly parameter. Its feedback amount is controlled by the Fl StatFB. Separate mix levels for the LFO taps and the static tap are then controlled by the Fl StatLvl and Fl LFO Lvl controls. The feedback and level controls can polarity invert each signal be setting them to negative values.

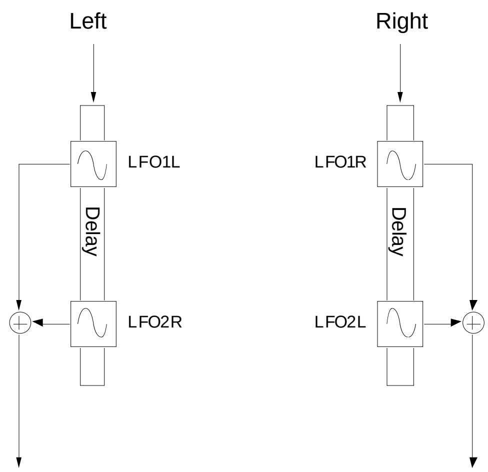


Figure 10-25 LFO delay taps in the configurable chorus and flange

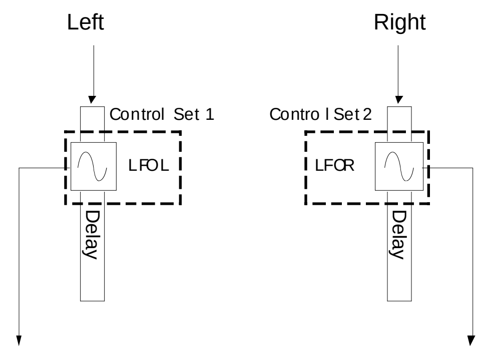


Figure 10-26 LFO control in Dual1Tap mode

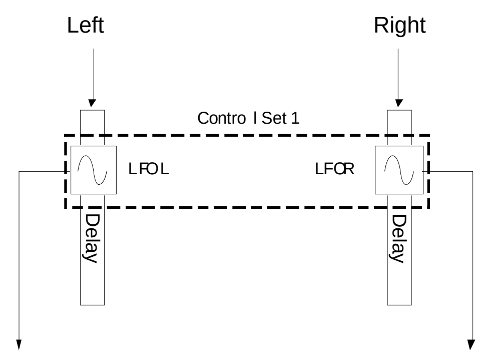


Figure 10-27 LFO control in Link1Tap mode

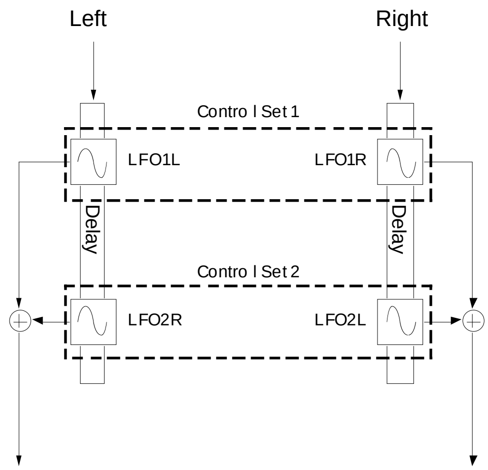


Figure 10-28 LFO control in Link2Tap mode

Parameters for Chorus

Page 1

Ch LFO cfg	Dual1Tap...	Ch LRPhase	0 to 360 deg
Ch Rate 1	0.01 to 10.00 Hz	Ch Rate 2	0.01 to 10.00 Hz
Ch Depth 1	0.0 to 100 ct	Ch Depth 2	0.0 to 100 ct
Ch Delay 1	0 to 1000 ms	Ch Delay 2	0 to 1000 ms
Ch Fdbk L	-100 to 100 %	Ch Fdbk R	-100 to 100 %
Ch Xcouple	0 to 100%	Ch HF Damp	16 to 25088 Hz

Parameters for Flange

Page 1

Fl LFO cfg	Dual1Tap...	Fl LRPhase	0 to 360 deg
Fl Rate 1	0.01 to 10.00 Hz	Fl Rate 2	0.01 to 10.00 Hz
Fl Xcurs 1	0 to 230 ms	Fl Xcurs 2	0 to 230 ms
Fl Delay 1	0 to 1000 ms	Fl Delay 2	0 to 1000 ms
Fl Fdbk 1	-100 to 100 %	Fl Fdbk 2	-100 to 100 %
Fl Phase 1	0 to 360 deg	Fl Phase 2	0 to 360 deg

Page 2

Fl HF Damp	16 to 25088 Hz
Fl Xcouple	0 to 100%
Fl StatDly	0 to 230 ms
Fl StatFB	-100 to 100 %
Fl StatLvl	-100 to 100 %
Fl LFO Lvl	-100 to 100 %

Ch LFO cfg	Sets the user interface mode for controlling each of the 4 chorus LFOs.
Ch LRPhase	Controls the relative phase between left channel LFOs and right channel LFOs. In Dual1Tap mode, however, this parameter is accurate only when Ch Rate 1 and Ch Rate 2 are set to the same speed, and only after the Ch LFO cfg parameter is moved, or the algorithm is called up.
Ch Fdbk L, Ch Fdbk R	These control the amount that the output of the chorus is fed back into the input.
All other Chorus parameters	Refer to Chorus documentation.
Fl LFO cfg	Sets the user interface mode for controlling each of the 4 flange LFOs.
Fl LRPhase	Controls the relative phase between left channel LFOs and right channel LFOs. In Dual1Tap mode, however, this parameter is accurate only when Fl Rate 1 and Fl Rate 2 are set to the same speed, and only after the Fl LFO cfg parameter is moved, or the algorithm is called up.
Fl Phase 1, Fl Phase 2	These adjust the corresponding LFO phase relationships between themselves and the internal beat clock.
All other Flange parameters	Refer to Flange documentation. Parameters with a 1 or 2 correspond to LFO taps organized as described above.

Laser Delay

Laser Delay is a tempo based delay with added functionality, including image shifting, cross-coupling, high frequency damping, low frequency damping, and a LaserVerb element. Separate left and right controls are provided for delay time, feedback, and laser controls. Parameters associated with Laser Verb in a combination algorithm begin with “Dly” or “Lsr”.

The delay length for each channel is determined by Dly Tempo, expressed in beats per minute (BPM), and the delay length (Dly Time L and Dly Time R) of each channel is expressed in beats (bts). The tempo alters both channel delay lengths together. With the tempo in beats per minute and delay lengths in beats, you can calculate the length of a delay in seconds as beats/tempo * 60 (sec/min). Since KDFX has a limited amount of delay memory available (usually 1.5 seconds for Laser Delay), selecting slow tempos and/or long delay lengths may cause you to run out of delay memory. At this point, each delay will pin at it’s maximum possible time. When you slow down the tempo, you may find the delays lose their sync.

The laser controls perform similarly to those found in LaserVerb, and affect the laser element of the effect. The LsrCntour changes the laser regeneration envelope shape. Higher values increase the regeneration amount, and setting it to 0% will disable the Laser Delay portion completely turning the effect into a basic delay. LsrSpace controls the impulse spacing of each regeneration. Low values create a strong initial pitched quality with slow descending resonances, while higher values cause the resonance to descend faster through each regeneration. See the LaserVerb section for more detailed information.

Delay regeneration is controlled collectively by the Dly Fdbk and LsrCntour parameters since the laser element contains feedback within itself. Setting both to 0% defeats all regeneration, including the laser element entirely. Increasing either one will increase regeneration overall, but with different qualities. Dly Fdbk is a feedback control in the classic sense, feeding the entire output of the effect back into the input, with negative values polarity inverting the signal. The LsrCntour parameter adds only the Laser Delay portion of the effect, including it’s own regeneration. For the most intense laser-ness, keep Dly Fdbk at 0% while LsrCntour is enabled.

Dly FBImag, Dly Xcouple, Dly HFDamp, and Dly LFDamp are just like those found in other algorithms. Not all Laser Delays in combination algorithms will have all four of these parameters due to resource allocation.

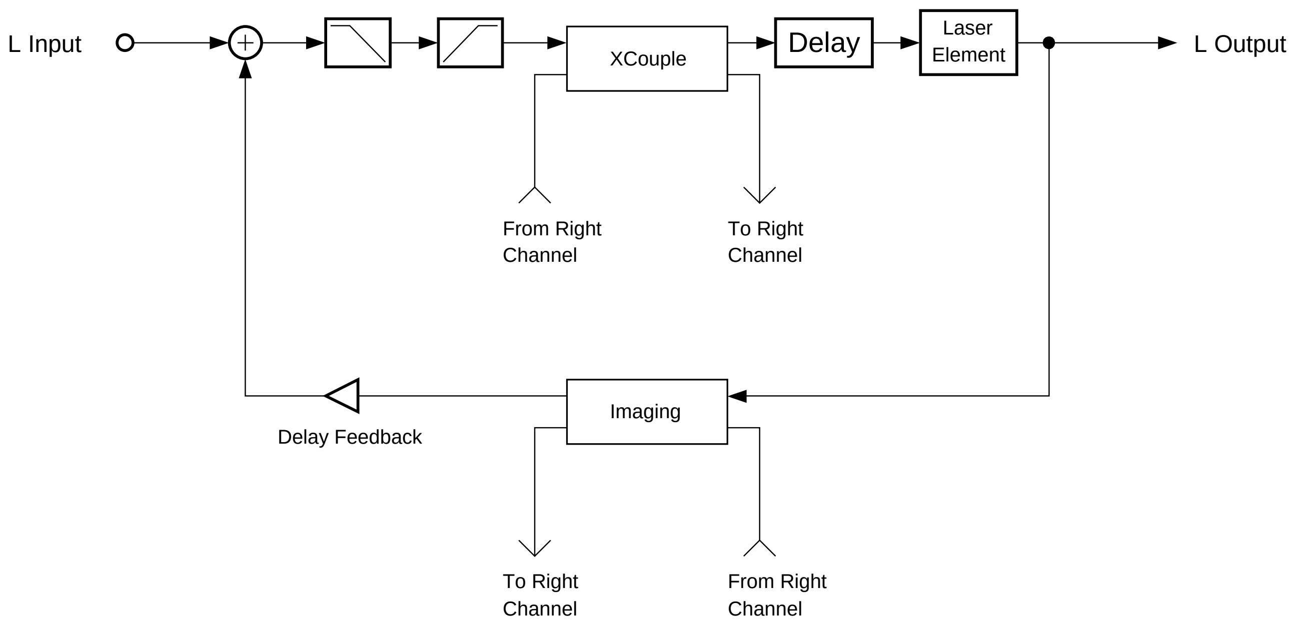


Figure 10-29 Laser Delay (left channel)

Parameters for Laser Delay

Dly Time L	0 to 6 bts	Dly Time R	0 to 6 bts
Dly Fdbk L	-100 to 100 %	Dly Fdbk R	-100 to 100 %
Dly HFDamp	0 to 32 bts	Dly FBImag	-100 to 100 %
Dly LFDamp	0.10 to 6.00 x	Dly Xcple	0 to 100%
LsrCntourL	0 to 100 %	LsrCntourR	0 to 100 %
LsrSpace L	0 to 100 samp	LsrSpace R	0 to 100 samp

Dly Time Left and Right. The delay lengths of each channel in beats. The duration of a beat is specified with the Tempo parameter. The delay length in seconds is calculated as beats/ tempo * 60 (sec/min).

Dly Fdbk Left and Right. The amount of the output of the effect that is fed back to the input.

Dly HFDamp Controls the cutoff frequency of a 1 pole (6dB/oct slope) lopass filter in the feedback path. The filter is heard when either Dly Fdbk or LsrCntour is used.

Dly LFDamp Controls the cutoff frequency of a 1 pole (6dB/oct slope) hipass filter in the feedback path. The filter is heard when either Dly Fdbk or LsrCntour is used.

Dly FBImag This parameter controls the amount of image shifting during each feedback regeneration, and is heard only when Dly Fdbk is used. Small positive values shift the image to the right, while small negative values shift to the left. Larger values tend to shift the image so far that the image gets scrambled, and in some cases create ambience.

Dly Xcple This parameter controls the amount of signal that is swapped between the left and right channels through each feedback generation when Dly Fdbk is used. A setting of 0% has no affect. 50% causes equal amounts of signal to be present in both channels causing the image to collapse into a center point source. A setting of 100% causes the left and right channels to swap each regeneration, which is also referred to as “ping-ponging”. The regeneration affects of cross-coupling are not heard when LsrCntour is used by itself.

LsrCntour Left and Right. Controls the overall envelope shape of the laser regeneration. When set to a high value, sounds passing through will start at a high level and slowly decay. As the control value is reduced, it takes some time for the effect to build up before decaying. When the Contour is set to zero, the laser portion is turned off turning regeneration into straight feedback.

LsrSpace Left and Right. Determines the starting pitch of the descending resonance and how fast it descends. See the section on Laser Delay for more detailed information.

Combination 4-Tap

Combination 4-Tap is a tempo based 4 tap delay with feedback used in combination algorithms. Parameters associated with the 4 tap effect start with “4T”. The control over the feedback tap and individual output taps is essentially the same as the 4-Tap Delay BPM algorithm, with the exception that the delay times will pin at the maximum delay time instead of automatically cutting their times in half. Additionally, the feedback path may also offer cross-coupling, an imager, a hipass filter, and/or a lopass filter.

Parameters for Combination 4-Tap

Page 1

4T LoopLen	0 to 32 bts
4T FB Lvl	-100 to 100 %
4T FB Imag	-100 to 100 %
4T FB XCpl	0 to 100 %
4T HF Damp	16 to 25088 Hz
4T LF Damp	16 to 25088 Hz

Page 2

Tap1 Delay	0 to 32 bts	Tap3 Delay	0 to 32 bts
Tap1 Level	-100 to 100 %	Tap3 Level	-100 to 100 %
Tap1 Bal	-100 to 100 %	Tap3 Bal	-100 to 100 %
Tap2 Delay	0 to 32 bts	Tap4 Delay	0 to 32 bts
Tap2 Level	-100 to 100 %	Tap4 Level	-100 to 100 %
Tap2 Bal	-100 to 100 %	Tap4 Bal	-100 to 100 %

4T FB Imag This parameter controls the amount of image shifting during each feedback regeneration. Small positive values shift the image to the right, while small negative values shift to the left. Larger values tend to shift the image so far that the image gets scrambled, and in some cases create ambience.

4T FB Xcpl This parameter controls the amount of signal that is swapped between the left and right channels through each feedback regeneration. A setting of 0% has no affect. 50% causes equal amounts of signal to be present in both channels causing the image to collapse into a center point source. A setting of 100% causes the left and right channels to swap each regeneration, which is also referred to as "ping-ponging".

All other parameters Refer to 4-Tap Delay BPM documentation.

Reverb

The reverbs offered in these combination effects is MiniVerb. Information about it can be found in the MiniVerb documentation. Parameters associated with this reverb begin with Rv.

MiniVerb

Rv Type	Hall1		
Rv Time	0.5 to 30.0 s; Inf		
Rv DiffScl	0.00 to 2.00x	Rv Density	0.00 to 4.00x
Rv SizeScl	0.00 to 4.00x	Rv HF Damp	16 to 25088 Hz
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms

Pitcher

The pitchers offered in these effects are the same as that found in its stand alone version. Review the Pitcher section for more information. Parameters associated with this effect begin with Pt.

Parameters for Pitcher

Pt Pitch	C-1 to G9
Pt Offset	-12.0 to 12.0 ST
Pt Odd Wts	-100 to 100 %
Pt PairWts	-100 to 100 %
Pt 1/4 Wts	-100 to 100 %
Pt 1/2 Wts	-100 to 100 %

Shaper

The shaper offered in these combination effects have the same sonic qualities as those found in VAST. Refer to the section on shapers in the *Musician's Guide* for an overview. Parameters associated with this effect begin with Shp.

This KDFX shaper also offers input and output 1 pole (6dB/oct) lopass filters controlled by the Shp Inp LP and Shp Out LP respectively. There is an additional output gain labeled Shp OutPad to compensate for the added gain caused by shaping a signal.

Parameters for Shaper

Shp Inp LP	16 to 25088 Hz
Shp Amt	0.10 to 6.00 x
Shp Out LP	16 to 25088 Hz
Shp OutPad	Off; -79.0 to 0.0 dB

Shp Inp LP	Adjusts the cutoff frequency of the 1 pole (6dB/oct) lopass filter at the input of the shaper.
Shp Out LP	Adjusts the cutoff frequency of the 1 pole (6dB/oct) lopass filter at the output of the shaper.
Shp Amount	Adjusts the shaper intensity. This is exactly like the one in VAST.
Shp OutPad	Adjusts the output gain at the output of the shaper to compensate for added gain caused by the shaper.

714 Quantize+Flange

Digital quantization followed by flanger

PAUs: 1

Digital audio engineers will go to great lengths to remove, or at least hide the effects of digital quantization distortion. In Quantize+Flange we do quite the opposite, making quantization an in-your-face effect. The quantizer will give your sound a dirty, grundgy, perhaps industrial sound. As you've already gathered from the name, the quantization is followed by a flanger. Quantize+Flange is a stereo effect.

Quantization distortion is a digital phenomenon caused by having only a limited number of bits with which to represent signal amplitudes (finite precision). You are probably aware that a bit is a number which can have only one of two values: 0 or 1. When we construct a data or signal word out of more than one bit, each additional bit will double the number of possible values. For example a two bit number can have one of four different values: 00, 01, 10 or 11. A three bit number can take one of eight different values, a four bit number can take one of sixteen values, etc. The 18 bits of the K2600's digital to analog converter (DAC) represents 262144 different amplitude levels (2^{18}). Let's take a look at how finite precision of digital words affects audio signals. The figures following are plots of a decaying sine wave with varying word lengths.

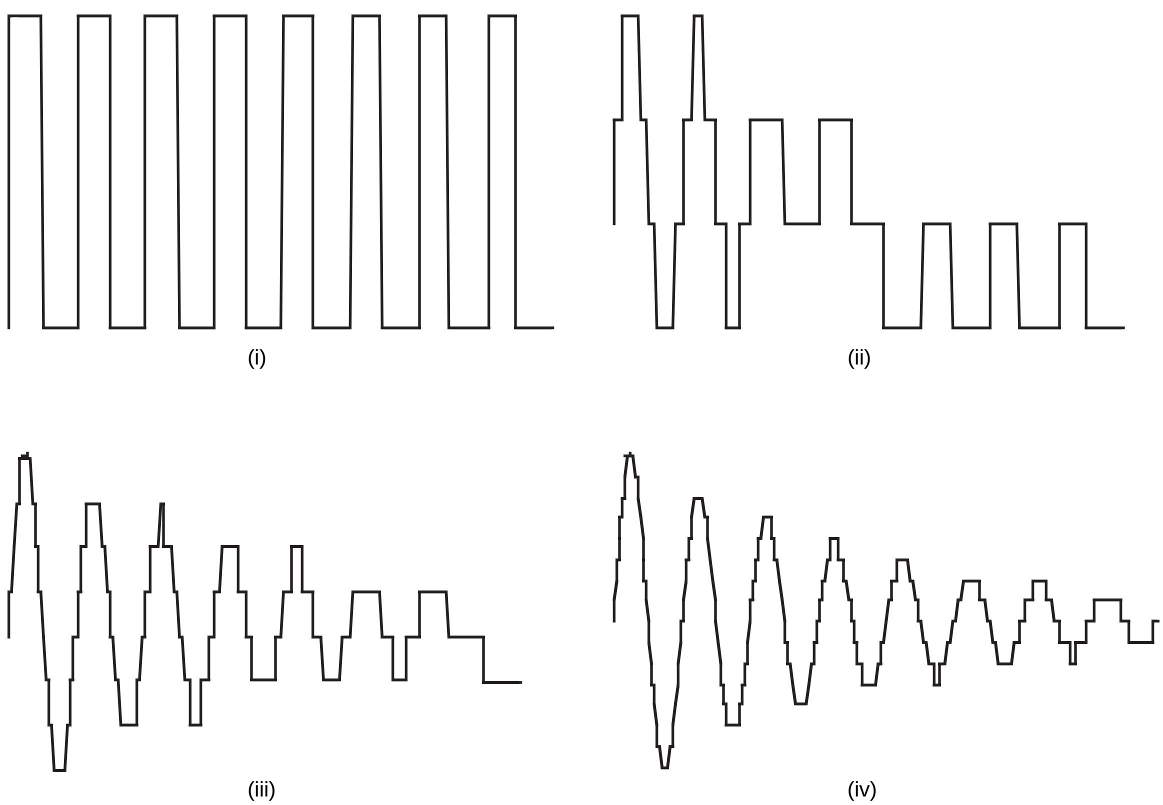


Figure 10-30 A decaying sine wave represented with different word lengths: (i) 1-bit, (ii) 2-bit, (iii) 3-bit, (iv) 4-bit.

Clearly a one bit word gives a very crude approximation to the original signal while four bits is beginning to do a good job of reproducing the original decaying sine wave. When a good strong signal is being quantized (its word length is being shortened), quantization usually sounds like additive noise. But notice that as the signal decays in the above figures, fewer and fewer quantization levels are being exercised until, like the one bit example, there are only two levels being toggled. With just two levels, your signal has become a square wave.

Controlling the bit level of the quantizer is done with the DynamRange parameter (dynamic range). A 0 dB we are at a one bit word length. Every 6 dB adds approximately one bit, so at 144 dB, the word length is 24 bits . The quantizer works by cutting the gain of the input signal, making the lowest bits fall off the end of the word. The signal is then boosted back up so we can hear it. At very low DynamRange settings, the step from one bit level to the next can become larger than the input signal. The signal can still make the quantizer toggle between bit level whenever the signal crosses the zero signal level, but with the larger bit levels, the output will get louder and louder. The Headroom parameter prevents this from happening. When the DynamRange parameter is lower than the Headroom parameter, no more signal boost is added to counter-act the cut used to quantize the signal. Find the DynamRange level at which the output starts to get too loud, then set Headroom to that level. You can then change the DynamRange value without worrying about changing the signal level. Headroom is a parameter that you set to match your signal level, then leave it alone.

At very low DynamRange values, the quantization becomes very sensitive to dc offset. It affects where your signal crosses the digital zero level. A dc offset adds a constant positive or negative level to the signal. By adding positive dc offset, the signal will tend to quantize more often to a higher bit level than to a lower bit level. In extreme cases (which is what we're looking for, after all), the quantized signal will sputter, as it is stuck at one level most of the time, but occasionally toggles to another level.

A flanger with one LFO delay tap and one static delay tap follows the quantizer. See the section on multi-tap flangers (Flanger1 and Flanger2) for a detailed explanation of how the flanger works.

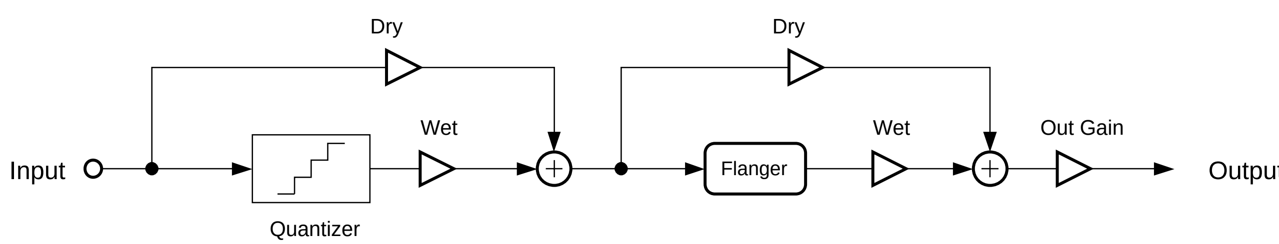


Figure 10-31 Block diagram of one channel of Quantize+Flange.

Quant W/D is a wet/dry control setting the relative amount of quantized (wet) and not quantized (dry) signals being passed to the flanger. The Flange W/D parameter similarly controls the wet/dry mix of the flanger. The dry signal for the flanger is the wet/dry mix output from the quantizer.

Parameters for Quantize + Flange

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Quant W/D	0 to 100%	DynamRange	0 to 144 dB
Flange W/D	-100 to 100%	dc Offset	-79.0 to 0.0 dB
		Headroom	0 to 144 dB

Page 2

Fl Tempo	System, 1 to 255 BPM	Fl Fdbk	-100 to 100%
Fl Period	0 to 32 bts		
Fl L Phase	0.0 to 360.0 deg	Fl R Phase	0.0 to 360.0 deg
Fl StatLvl	-100 to 100%	Fl LFO Lvl	-100 to 100%

Page 3

FlStatDlyC	0.0 to 230.0 ms	Fl Xcurs C	0.0 to 230.0 ms
FlStatDlyF	-127 to 127 samp	Fl Xcurs F	-127 to 127 samp
		Fl Delay C	0.0 to 230.0 ms
		Fl Delay F	-127 to 127 samp

In/Out When set to “In”, the quantizer and flanger are active; when set to “Out”, the quantizer and flanger are bypassed.

Out Gain The overall gain or amplitude at the output of the effect.

Quant W/D The relative amount of quantized (wet) to unaffected (dry) signal passed to the flanger. At 100%, you hear only quantized signal pass to the flanger.

Flange W/D The relative amount of input signal (from the quantizer) and flanger signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the quantizer (dry). When set to 100%, the output is all wet. Negative values polarity invert the wet signal.

DynamRange The digital dynamic range controls signal quantization, or how many bits to remove from the signal data words. At 0 dB the hottest of signals will toggle between only two bit (or quantization) levels. Every 6 dB added doubles the number of quantization levels. If the signal has a lot of headroom (available signal level before digital clipping), then not all quantization levels will be reached.

Headroom When the signal has a lot of headroom (available signal level before digital clipping), turning down DynamRange can cause the amplitude of adjacent quantization levels to exceed the input signal level. This causes the output to get very loud. Set Headroom to match the amount of digital signal level still available (headroom). This is easily done by finding the DynamRange level at which the signal starts getting louder and matching Headroom to that value.

dc Offset Adds a positive dc Offset to the input signal. By adding dc Offset, you can alter the position where digital zero is with respect to you signal. At low DynamRange settings, adding dc Offset can may the output sputter. dc Offset is expressed in decibels (dB) relative to full scale digital.

Fl Tempo Basis for the rates of the LFOs, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

Fl Period Sets the LFO rate based on the Tempo determined above: the number of beats corresponding to one period of the LFO cycle. For example, if the Fl Period is set to “4”, the LFOs will take four beats to pass through one oscillation, so the LFO rate will be 1/4th of the Tempo setting. If it is set to “6/24” (=1/4), the LFO will oscillate four times as fast as the Tempo. At “0”, the LFOs stop oscillating and their phase is undetermined (wherever they stopped).

Fl Fdbk The level of the flanger feedback signal into the flanger delay line. The feedback signal is taken from the LFO delay tap. Negative values polarity invert the feedback signal.

Fl L/R Phase The phase angles of the left and right LFOs relative to each other and to the system tempo clock, if turned on (see Fl Tempo). In all other respects the right and left channels are symmetric. For example, if one LFO is set to 0° and another is set to 180°, then when one LFO delay tap is at its shortest, the other will be at its longest. If the system tempo clock is on, the LFOs are synchronized to the clock with absolute phase. A phase of 0° will put an LFO tap at the center of its range and its lengthening. Using different phase angles for left and right, the stereo sound field is broken up and a stereo image becomes difficult to spatially locate. The effect is usually described as “phasey”. It tends to impart a greater sense of motion.

Fl StatLvl The level of the flanger static delay tap. Negative values polarity invert the signal. Setting the tap level to 0% turns off the delay tap.

Fl LFO Lvl The level of the flanger LFO modulated delay tap. Negative values polarity invert the signal. Setting the tap level to 0% turns off the delay tap.

FlStatDlyC The nominal length of the flanger static delay tap from the delay input. The name suggests the tap is stationary, but it can be connected to a control source such as a data slider, a ribbon, or a V.A.S.T. function to smoothly vary the delay length. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective.

FlStatDlyF A fine adjustment to the flanger static delay tap length. The resolution is one sample.

Fl Xcurs C The flanger LFO excursion controls set how far the LFO modulated delay taps can move from the center of their ranges. The total range of the LFO sweep is twice the excursion. If the excursion is set to 0, the LFO does not move and the tap behaves like a simple delay line set to the minimum delay. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. This parameter is a coarse adjustment for the excursion.

Fl Xcurs F A fine adjustment for the flanger LFO excursions. The resolution is one sample.

Fl Delay C The minimum delay for the flanger LFO modulated delay taps. The maximum delay will be the minimum plus twice the excursion. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. This parameter is a coarse adjustment for the delay.

Fl Delay F A fine adjustment to the minimum flanger delay tap lengths. The resolution is one sample.

715 Dual MovDelay
716 Quad MovDelay

Generic dual mono moving delay lines

PAUs: 1 for Dual
2 for Quad

Each of these algorithms offers generic moving delay lines in a dual mono configuration. Each separate moving delay can be used as a flanger, chorus, or static delay line selectable by the LFO Mode parameter. Both flavors of chorus pitch envelopes are offered: ChorTri for triangle, and ChorTrap for trapezoidal pitch shifting. Refer to the Chorus section for more information on these envelope shapes.

The value functions much like a wet/dry mix where 0% means that only the algorithm input dry signal is fed into effect B (putting the effects in parallel), and 100% means only the output of effect A is fed into effect B (putting the effects in series). See Figure 1 for signal flow of Chorus+4Tap as an example.

720 MonoPitcher+Chor
721 MonoPitcher+Flan

Mono pitcher algorithm (filter with harmonically related resonant peaks) with a chorus or flanger

PAUs: 2 each

The mono pitcher algorithm applies a filter which has a series of peaks in the frequency response to the input signal. The peaks may be adjusted so that their frequencies are all multiples of a selectable frequency, all the way up to 24 kHz. When applied to a sound with a noise-like spectrum (white noise, with a flat spectrum, or cymbals, with a very dense spectrum of many individual components), an output is produced which sounds very pitched, since most of its spectral energy ends up concentrated around multiples of a fundamental frequency.

The graphs below show Pt PkSplit going from 0% to 100%, for a Pt Pitch of 1 khz (approx. C6), and Pt PkShape set to 0.

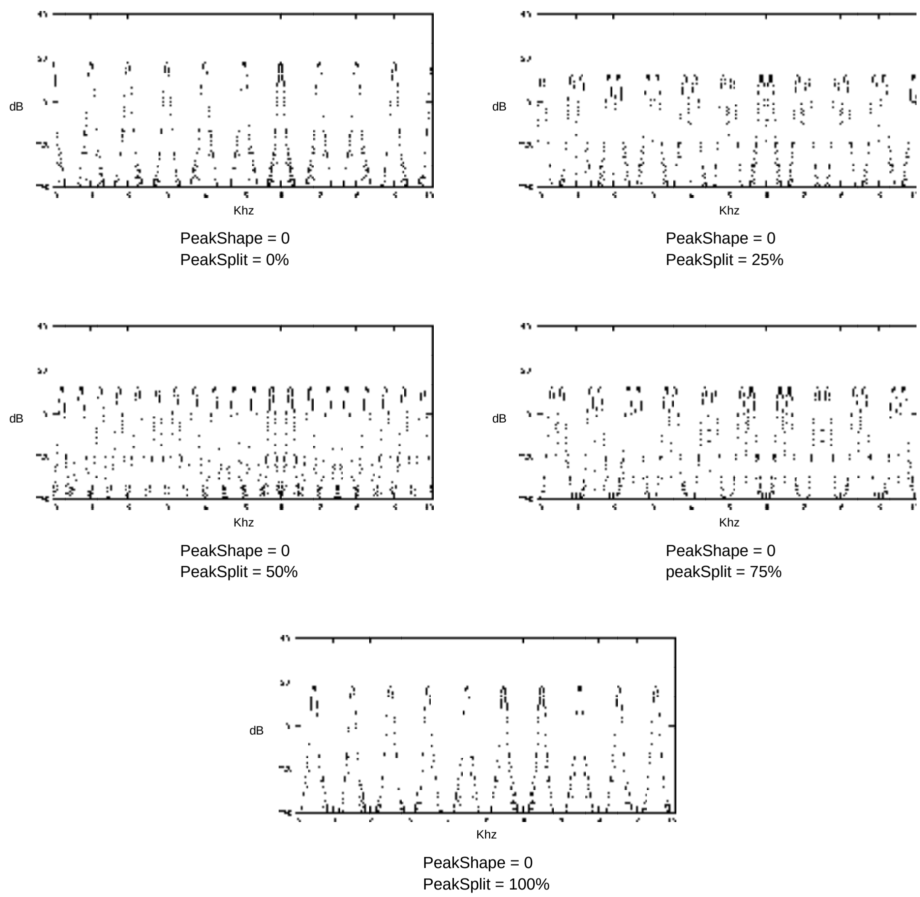
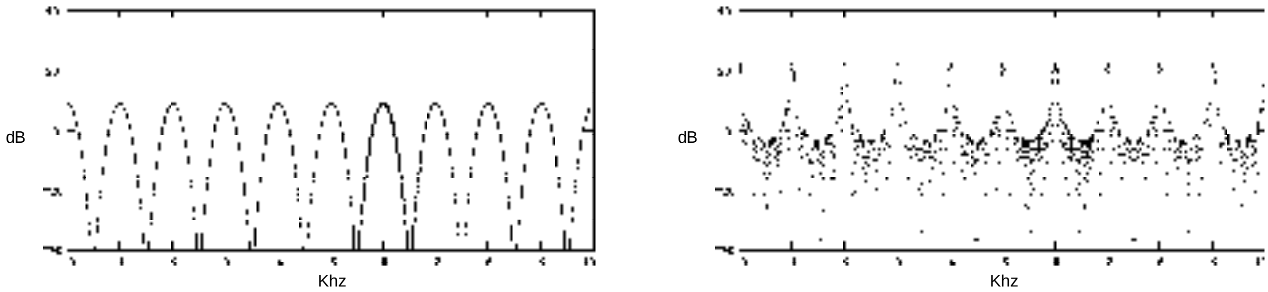


Figure 10-32 Response of Pitcher with different PkSplit settings. Pitch is C6 and PkShape is 0.

Note that a Pt PkSplit of 100% gives only odd multiples of a fundamental that is one octave down from no splitting. The presence of only odd multiples will produce a hollow sort of sound, like a square wave (which also only has odd harmonics.) Curiously enough, at a Pt PkSplit of 50% we also get odd multiples of a frequency that is now two octaves below the original Pitch parameter. In general, most values of PkSplit will give peak positions that are not harmonically related.

The figures below show Pt PkShape of -1.0 and 1.0, for a Pitch of C6 and a PkSplit of 0%.



PeakShape = 1.0
PeakSplit = 0%

PeakShape = -1.0
PeakSplit = 0%

Figure 10-33 Response of Pitcher with different PkShape settings.

Applying Pitcher to sounds such as a single sawtooth wave will tend to not produce much output, unless the sawtooth frequency and the Pitcher frequency match or are harmonically related, because otherwise the peaks in the input spectrum won't line up with the peaks in the Pitcher filter. If there are enough peaks in the input spectrum (obtained by using sounds with noise components, or combining lots of different simple sounds, especially low pitched ones, or severly distorting a simple sound) then Pitcher can do a good job of imposing its pitch on the sound.

Multiple Pitcher algorithms can be run (yes, it takes all of KDFX to get three) to produce chordal output.

A vocoder-like effect can be produced, although in some sense it works in exactly an opposite way to a real vocoder. A real vocoder will superimpose the spectrum of one signal (typically speech) onto a musical signal (which has only a small number of harmonically related spectral peaks.) Pitcher takes an input such as speech, and then picks out only the components that match a harmonic series, as though they were from a musical note.

Configurable Flange

The flange in alg 721 is a configurable flange. Refer to the section on Configurable Chorus and Flange for details about this effect.

Chorus

The chorus used in alg 720 is a basic dual channel chorus. Refer to Chorus documentation for more information on the effect.

Parameters for MonoPitcher + Chor

Page 1

Wet/Dry	100 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Mix Pitchr	-100 to 100%		
Mix Chorus	-100 to 100%		
Pt/Dry->Ch	0 to 100%		

Page 2

Pt Inp Bal	-100 to 100%	Pt Out Pan	-100 to 100%
Pt Pitch	C-1 to G 9	Pt Offset	-12.0 to 12.0 ST
Pt PkSplit	0 to 100%	Pt PkShape	-1.0 to 1.0

Page 3

ChPtchEnvL	Triangle or Trapzoid	ChPtchEnvL	Triangle or Trapzoid
Ch Rate L	0.01 to 10.00 Hz	Ch Rate R	0.01 to 10.00 Hz
Ch Depth L	0.0 to 100.0 ct	Ch Depth R	0.0 to 100.0 ct
Ch Delay L	0.0 to 720.0 ms	Ch Delay R	0.0 to 720.0 ms
Ch Fdbk L	-100 to 100%	Ch Fdbk R	-100 to 100%
Ch Xcouple	0 to 100%	Ch HF Damp	16 to 25088 Hz

Parameters for MonoPitcher + Flan

Page 1

Wet/Dry	100 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Mix Pitchr	-100 to 100%		
Mix Flange	-100 to 100%	Fl Tempo	System, 1 to 255 BPM
Pt/Dry->Fl	0 to 100%		

Page 2

Pt Inp Bal	-100 to 100%	Pt Out Pan	-100 to 100%
Pt Pitch	C-1 to G 9	Pt Offset	-12.0 to 12.0 ST
Pt PkSplit	0 to 100%	Pt PkShape	-1.0 to 1.0

Page 3

Fl LFO cfg	Dual1Tap	Fl LRPhase	0.0 to 360.0 deg
Fl Rate 1	0 to 32 bts	Fl Rate 2	0 to 32 bts
Fl Xcurs 1	0.0 to 230.0 bts	Fl Xcurs 2	0.0 to 230.0 bts
Fl Delay 1	0.0 to 230.0 ms	Fl Delay 2	0.0 to 230.0 ms
Fl Phase 1	0.0 to 360.0 deg	Fl Phase 2	0.0 to 360.0 deg
Fl Fdbk	-100 to 100%	Fl HF Damp	16 to 25088 Hz

Wet/Dry This is a simple mix of the pitched and chorused or flanged signal relative to the dry input signal.

Out Gain The overall gain or amplitude at the output of the effect.

Mix Pitchr The amount of the pitcher signal to be sent directly to the output as a percent. Any signal that this parameter sends to the output does not get sent to the chorus or flanger.

Mix Chorus, Mix Flange	The amount of the flanger or chorus signal to send to the output as a percent.
Pt/Dry->Ch, Pt/Dry->Fl	The relative amount of pitcher signal to dry signal to send to the chorus or flanger. At 0% the dry input signal is routed to the chorus or flanger. At 100%, the chorus or flanger receives its input entirely from the pitcher.
Pt Inp Bal	Since this is a mono algorithm, an input balance control is provided to mix the left and right inputs to the pitcher. -100% is left only, 0% is left plus right, and 100% is right only.
Pt Out Pan	Pans the mono pitcher output from left (-100%) to center (0%) to right (100%)
Pt Pitch	The "fundamental" frequency of the Pitcher output. This sets the frequency of the lowest peak in terms of standard note names. All the other peaks will be at multiples of this pitch.
Pt PkSplit	Splits the pitcher peaks into two peaks, which both move away from their original unsplit position, one going up and the other down in frequency. At 0% there is no splitting; all peaks are at multiples of the fundamental. At 100% the peak going up merges with the peak going down from the next higher position.
Pt Offset	An offset in semitones from the frequency specified in Pitch.
Pt PkShape	Controls the shape of the pitcher spectral peaks. 0.0 gives the most "pitchiness" to the output, in that the peaks are narrow, with not much energy between them. -1.0 makes the peaks wider. 1.0 brings up the level between the peaks.
All other Chorus parameters	Refer to Chorus documentation.
Fl LFO cfg	Sets the user interface mode for controlling each of the 4 flange LFOs.
Fl LRPhase	Controls the relative phase between left channel LFOs and right channel LFOs. In Dual1Tap mode, however, this parameter is accurate only when Fl Rate 1 and Fl Rate 2 are set to the same speed, and only after the Fl LFO cfg parameter is moved, or the algorithm is called up.
Fl Phase 1, Fl Phase 2	These adjust the corresponding LFO phase relationships between themselves and the internal beat clock.
All other Flange parameters	Refer to Flange documentation. Parameters with a 1 or 2 correspond to LFO taps organized as described above.

Distortion Algorithms

724 Mono Distortion
725 MonoDistort + Cab
726 MonoDistort + EQ
728 StereoDistort+EQ

Small distortion algorithms

PAUs: 1 for Mono Distortion
2 for MonoDistort + Cab
2 for MonoDistort + EQ
3 for StereoDistort + EQ

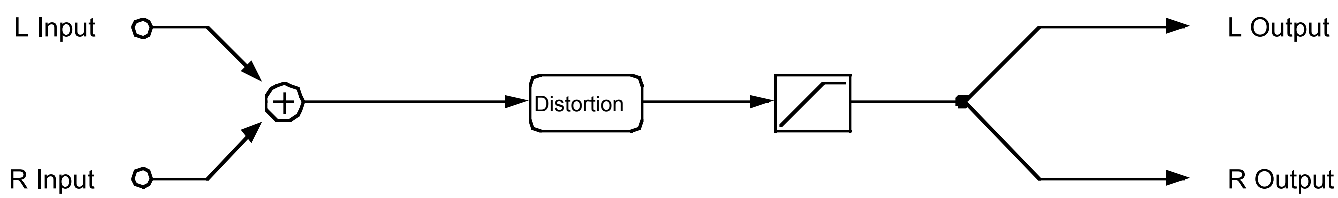


Figure 10-34 Block diagram of Mono Distortion

Mono Distortion sums its stereo input to mono, performs distortion followed by a highpass filter and sends the result as centered stereo.

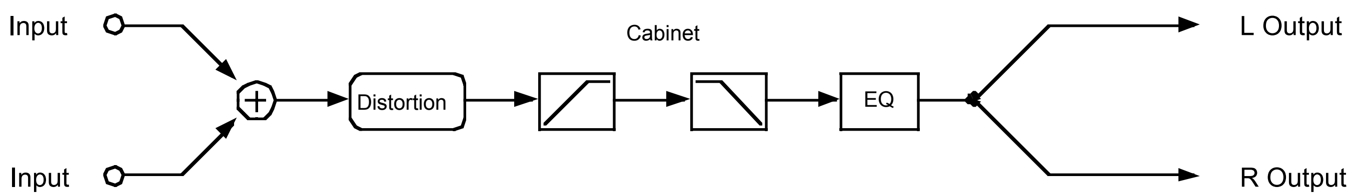


Figure 10-35 Block diagram of MonoDistort + EQ

MonoDistort + EQ is similar to Mono Distortion except the single highpass filter is replaced with a pair of second-order highpass/lowpass filters to provide rudimentary speaker cabinet modeling. The highpass and lowpass filters are then followed by an EQ section with bass and treble shelf filters and two parametric mid filters.

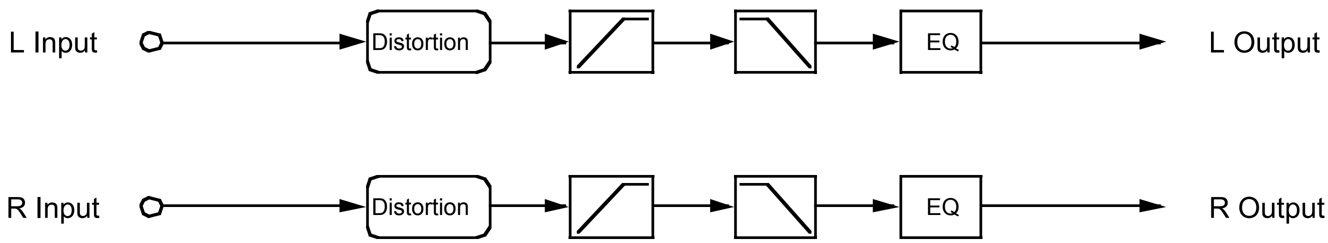


Figure 10-36 Block diagram of StereoDistort+EQ

StereoDistort + EQ processes the left and right channels separately, though there is only one set of parameters for both channels. The stereo distortion has only 1 parametric mid filter.

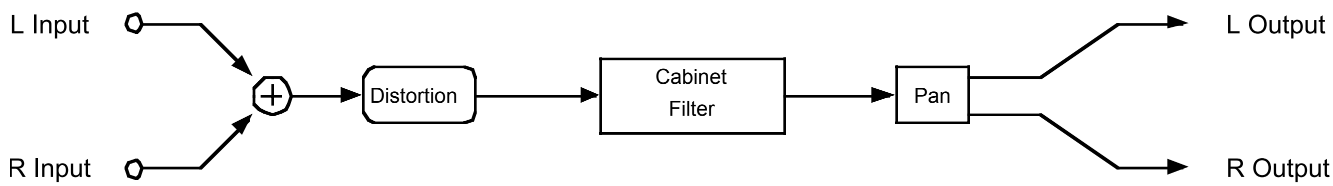


Figure 10-37 Block diagram of MonoDistort + Cab

MonoDistort + Cab is also similar to Mono Distortion except the highpass is replaced by a full speaker cabinet model. There is also a panner to route the mono signal between left and right outputs. In MonoDistort + Cab, the distortion is followed by a model of a guitar amplifier cabinet. The model can be bypassed, or there are 8 presets which were derived from measurments of real cabinets.

The distortion algorithm will soft clip the input signal. The amount of soft clipping depends on how high the distortion drive parameter is set. Soft clipping means that there is a smooth transition from linear gain to saturated overdrive. Higher distortion drive settings cause the transition to become progressively sharper or “harder”. The distortion never produces hard or digital clipping, but it does approach it at high drive settings. When you increase the distortion drive parameter you are increasing the gain of the algorithm until the signal reaches saturation. You will have to compensate for increases in drive gain by reducing the output gain. These algorithm will not digitally clip unless the output gain is over-driven.

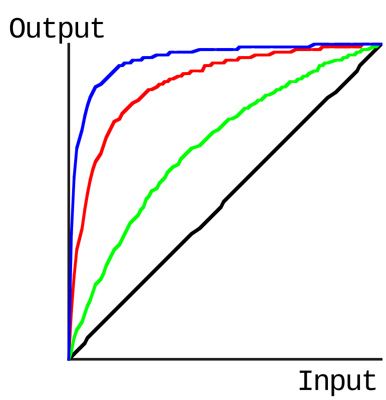


Figure 10-38 Input/Output Transfer Characteristic of Soft Clipping at Various Drive Settings

Signals that are symmetric in amplitude (they have the same shape if they are inverted, positive for negative) will usually produce odd harmonic distortion. For example, a pure sine wave will produce smaller copies of itself at 3, 5, 7, etc. times the original frequency of the sine wave. In the MonoDistort + EQ, a dc offset may be added to the signal to break the amplitude symmetry and will cause the distortion to produce even harmonics. This can add a "brassy" character to the distorted sound. The dc offset added prior to distortion gets removed at a later point in the algorithm.

Parameters for Mono Distortion

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	0 to 96 dB		
Warmth	16 to 25088 Hz		
Highpass	16 to 25088 Hz		

Parameters for MonoDistort + Cab

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	0 to 96 dB		
Warmth	16 to 25088 Hz	Cab Bypass	In or Out
		Cab Preset	Plain

Parameters for MonoDistort + EQ

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	0 to 96 dB		
Warmth	16 to 25088 Hz	dc Offset	-100 to 100%
Cabinet HP	16 to 25088 Hz	Cabinet LP	16 to 25088 Hz

Page 2

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	16 to 25088 Hz	Treb Freq	16 to 25088 Hz
Mid1 Gain	-79.0 to 24.0 dB	Mid2 Gain	-79.0 to 24.0 dB
Mid1 Freq	16 to 25088 Hz	Mid2 Freq	16 to 25088 Hz
Mid1 Width	0.010 to 5.000 oct	Mid2 Width	0.010 to 5.000 oct

Parameters for StereoDistort + EQ

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	0 to 96 dB		
Warmth	16 to 25088 Hz		
Cabinet HP	16 to 25088 Hz	Cabinet LP	16 to 25088 Hz

Page 2

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	16 to 25088 Hz	Treb Freq	16 to 25088 Hz
Mid Gain	-79.0 to 24.0 dB		
Mid Freq	16 to 25088 Hz		
Mid Width	0.010 to 5.000 oct		

Wet/Dry The amount of distorted (wet) signal relative to unaffected (dry) signal.

Out Gain The overall gain or amplitude at the output of the effect. For distortion, it is often necessary to turn the output gain down as the distortion drive is turned up.

Dist Drive Applies a boost to the input signal to overdrive the distortion algorithm. When overdriven, the distortion algorithm will soft-clip the signal. Since distortion drive will make your signal very loud, you may have to reduce the Out Gain as the drive is increased.

Warmth A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal.

Cab Bypass The guitar amplifier cabinet simulation may be bypassed. When set to “In”, the cabinet simulation is active; when set to “Out”, there is no cabinet filtering. [MonoDistort + Cab]

Cab Preset Eight preset cabinets have been created based on measurements of real guitar amplifier cabinets. The presets are Plain, Lead 12, 2x12, Open 12, Open 10, 4x12, Hot 2x12, and Hot 12. [MonoDistort + Cab]

Highpass Allows you to reduce the bass content of the distortion content. If you need more filtering to better simulate a speaker cabinet, you will have to choose a larger distortion algorithm. [Mono Distortion]

Cabinet HP A highpass filter which controls the low frequency limit of a simulated loudspeaker cabinet. [MonoDistort + EQ and StereoDistort+EQ]

Cabinet LP A lowpass filter which controls the high frequency limit of a simulated loudspeaker cabinet. [MonoDistort + EQ and StereoDistort+EQ]

Bass Gain The amount of boost or cut that the bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency. [MonoDistort + EQ and StereoDistort+EQ]

Bass Freq The center frequency of the bass shelving filter in intervals of one semitone. [MonoDistort + EQ and StereoDistort+EQ]

Treb Gain The amount of boost or cut that the treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency. [MonoDistort + EQ and StereoDistort+EQ]

Treb Freq The center frequency of the treble shelving filter in intervals of one semitone. [MonoDistort + EQ and StereoDistort+EQ]

Mid Gain The amount of boost or cut that the mid parametric filter should apply in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency. [MonoDistort + EQ and StereoDistort+EQ]

Mid Freq The center frequency of the mid parametric filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency. [MonoDistort + EQ and StereoDistort+EQ]

Mid Wid The bandwidth of the mid parametric filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response. [MonoDistort + EQ and StereoDistort+EQ]

727 PolyDistort + EQ

Eight stage distortion followed by equalization

PAUs: 2

PolyDistort + EQ is a distortion algorithm followed by equalization. The algorithm consists of an input gain stage, and then eight cascaded distortion stages. Each stage is followed by a one pole LP filter. There is also a one pole LP in front of the first stage. After the distortion there is a 4 band EQ section: Bass, Treble, and two Parametric Mids.

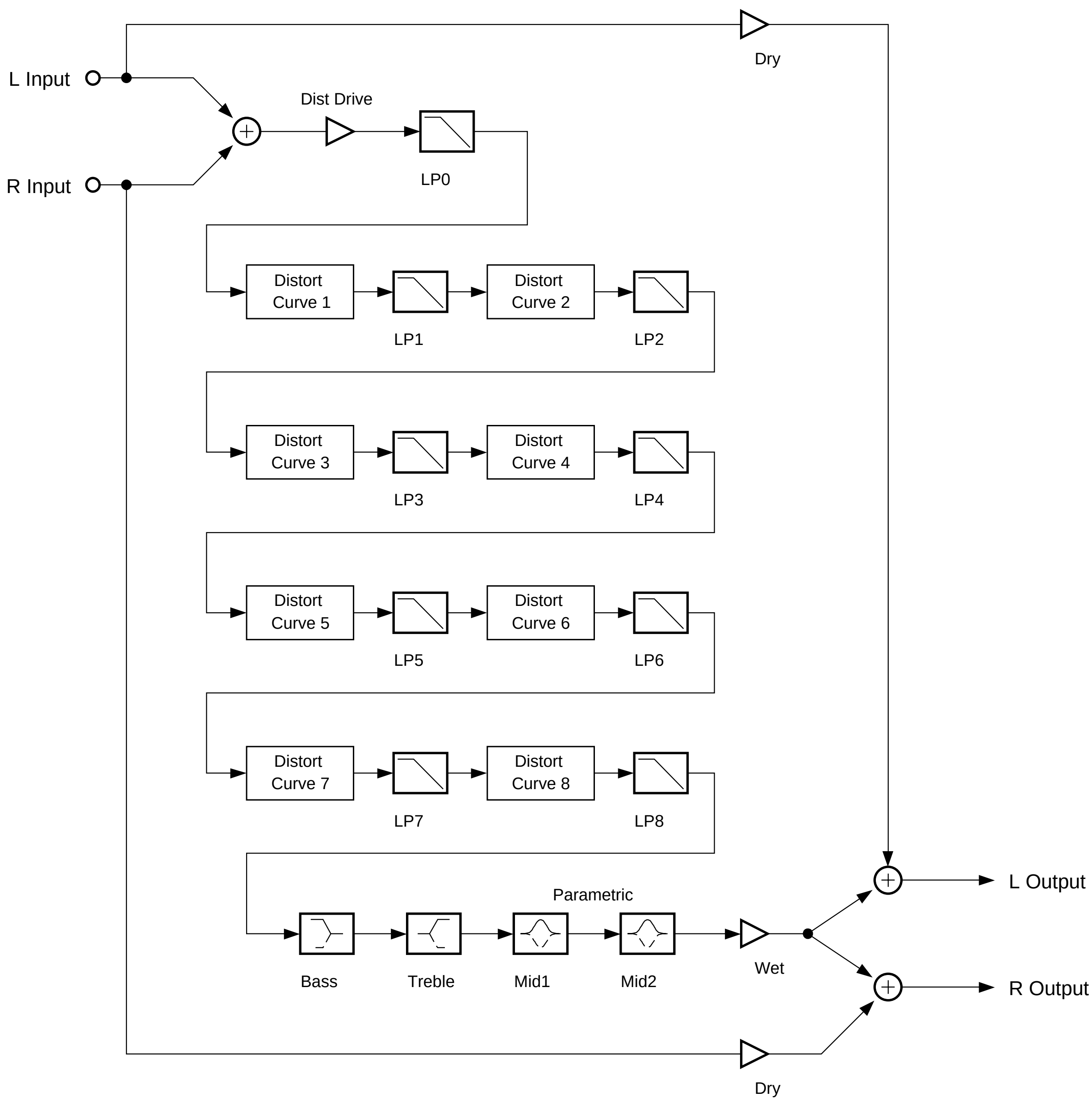


Figure 10-39 Block diagram of PolyDistort + EQ

PolyDistort is an unusual distortion algorithm which provides a great number of parameters to build a distortion sound from the ground up. The eight distortion stages each add a small amount of distortion to your sound. Taken together, you can get a very harsh heavy metal sound. Between each distortion stage is a low pass filter. The low pass filters work with the distortion stages to help mellow out the sound. Without any low pass filters the distortion will get very harsh and raspy.

Stages of distortion can be removed by setting the Curve parameter to 0. You can then do a 6, 4, or 2 stage distortion algorithm. The corresponding low passes should be turned off if there is no distortion in a section. More than 4 stages seem necessary for lead guitar sounds. For a cleaner sound, you may want to limit yourself to only 4 stages.

Once you have set up a distorted sound you are satisfied with, the Dist Drive parameter controls the input gain to the distortion, providing a single parameter for controlling distortion amount. You will probably find that you will have to cut back on the output gain as you drive the distortion louder.

Post distortion EQ is definitely needed for make things sound right. This should be something like a guitar speaker cabinet simulator, although not exactly, since we are already doing a lot of low pass filtering inside the distortion itself. Possible EQ settings you can try are Treble -20 dB at 5 Khz, Bass -6 dB at 100 Hz, Mid1, wide, +6 dB at 2 kHz, Mid2, wide, +3 dB at 200 Hz, but of course you should certainly experiment to get your sound. The Treble is helping to remove raspiness, the Bass is removing the extreme low end like an open-back guitar cabinet (not that guitar speaker have that much low end anyway), Mid1 adds enough highs so that things can sound bright even in the presence of all the HF roll-off, and Mid2 adds some warmth. Your favorite settings will probably be different. Boosting the Treble may not be a good idea.

Pre distortion EQ, available on the Studio INPUT page, is also useful for shaping the sound. EQ done in front of the distortion will not be heard as simple EQ, because the distortion section makes an adjustment in one frequency range felt over a much wider range due to action of the distortion. Simple post EQ is a bit too obvious for the ear, and it can get tired of it after a while.

Parameters for PolyDistort + EQ

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	Off, -79.0 to 48.0 dB		

Page 2

Curve 1	0 to 127%	Curve 5	0 to 127%
Curve 2	0 to 127%	Curve 6	0 to 127%
Curve 3	0 to 127%	Curve 7	0 to 127%
Curve 4	0 to 127%	Curve 8	0 to 127%

Page 3

LP0 Freq	16 to 25088 Hz		
LP1 Freq	16 to 25088 Hz	LP5 Freq	16 to 25088 Hz
LP2 Freq	16 to 25088 Hz	LP6 Freq	16 to 25088 Hz
LP3 Freq	16 to 25088 Hz	LP7 Freq	16 to 25088 Hz
LP4 Freq	16 to 25088 Hz	LP8 Freq	16 to 25088 Hz

Page 4

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	16 to 25088 Hz	Treb Freq	16 to 25088 Hz
Mid1 Gain	-79.0 to 24.0 dB	Mid2 Gain	-79.0 to 24.0 dB
Mid1 Freq	16 to 25088 Hz	Mid2 Freq	16 to 25088 Hz
Mid1 Width	0.010 to 5.000 oct	Mid2 Width	0.010 to 5.000 oct

Wet/Dry This is a simple mix of the distorted signal relative to the dry undistorted input signal.

Out Gain The overall gain or amplitude at the output of the effect. For distortion, it is often necessary to turn the output gain down as the distortion drive is turned up.

Dist Drive Applies gain to the input prior to distortion. It is the basic “distortion drive” control. Anything over 0 dB could clip. Normally clipping would be bad, but the distortion algorithm tends to smooth things out. Still, considering that for some settings of the other parameters you would have to back off the gain to -48 dB in order to get a not very distorted sound for full scale input, you should go easy on this amount.

Curve *n* The curvature of the individual distortion stages. 0% is no curvature (no distortion at all). At 100%, the curve bends over smoothly and becomes perfectly flat right before it goes into clipping.

LP *n* Freq These are the one pole low pass controls. LP0 Freq handles the initial low pass prior to the first distortion stage. The other low pass controls follow their respective distortion stages. With all low passes out of the circuit (set to the highest frequency), the sound tends to be too bright and raspy. With less distortion drive, less filtering is needed. If you turn off a distortion stage (set to 0%), you should turn of the low pass filter by setting it to the highest frequency.

Bass Gain The amount of boost or cut that the bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

Bass Freq The center frequency of the bass shelving filter in intervals of one semitone.

Treb Gain The amount of boost or cut that the treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

Treb Freq The center frequency of the treble shelving filter in intervals of one semitone.

Mid Gain The amount of boost or cut that the mid parametric filter should apply in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

Mid Freq The center frequency of the mid parametric filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

Mid Wid The bandwidth of the mid parametric filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

733 VibChor+Rotor 2
737 VibChor+Rotor 4

Vibrato/chorus into optional distortion into rotating speaker

PAUs: 2 for VibChor+Rotor 2
4 for VibChor+Rotor 4

The VibChor+Rotor algorithms contain multiple effects designed for the Hammond B3® emulation (KB3 mode). These effects are the Hammond® vibrato/chorus, amplifier distortion, and rotating speaker (Leslie®). Each of these effects may be turned off or bypassed, or the entire algorithm may be bypassed.

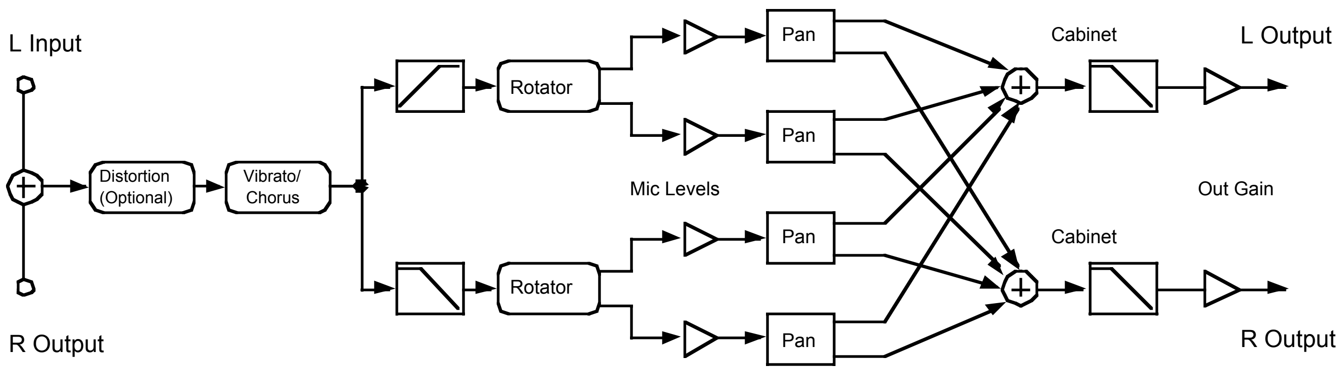


Figure 10-40 Block diagram of VibChor+Rotor

The first effect in the chain is the Hammond vibrato/chorus algorithm. The vibrato/chorus has six settings which are the same as those used in the Hammond B3: three vibrato (V1, V2, V3) and three chorus (C1, C2, C3) settings. In VibChor+Rotor 4, the vibrato chorus has been carefully modelled after the electro-mechanical vibrato/chorus in the B3. The vibrato/chorus in VibChor+Rotor 2 uses a conventional design, which has been set to match the B3 sound as closely as possible, but does not quite have the same character as the VibChor++Rotor 4 vibrato/chorus.

In VibChor+Roto 4 an amplifier distortion algorithm follows the vibrato/chorus. The distortion algorithm will soft clip the input signal. The amount of soft clipping depends on how high the distortion drive parameter is set. Soft clipping means that there is a smooth transition from linear gain to saturated overdrive. Higher distortion drive settings cause the transition to become progressively sharper or "harder". The distortion never produces hard or digital clipping, but it does approach it at high drive settings. When you increase the distortion drive parameter you are increasing the gain of the algorithm until the signal reaches saturation. You will have to compensate for increases in drive gain by reducing the output gain. These algorithm will not digitally clip unless the output gain is over-driven.

Finally the signal passes through a rotating speaker routine. The rotating speaker has separately controllable tweeter and woofer drivers. The signal is split into high and low frequency bands and the two bands are run through separate rotators. The upper and lower rotors each have a pair of virtual microphones which can be positioned at varying positions (angles) around the rotors. An angle of 0° is loosely defined as the front. You can also control the levels and left-right panning of each virtual microphone. The signal is then passed through a final lowpass filter to simulate the band-limiting effect of the speaker cabinet.

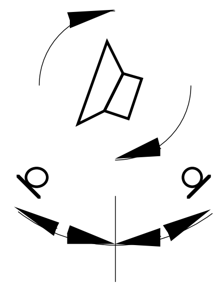


Figure 10-41 Rotating speaker with virtual microphones

For the rotating speakers, you can control the cross-over frequency of the high and low frequency bands (the frequency where the high and low frequencies get separated). The rotating speakers for the high and low frequencies have their own controls. For both, the rotation rate, the effective driver size and tremolo can be set. The rotation rate of course sets how fast the rotating speaker is spinning. The effective driver size is the radius of the path followed by the speaker relative to its center of rotation. This parameter is used to calculate the resulting Doppler shift of the moving speaker. Doppler shift is the pitch shift that occurs when a sound source moves toward or away from you the listener. In a rotating speaker, the Doppler shift will sound like vibrato. As well as Doppler shift, there will be some acoustic shadowing as the speaker is alternately pointed away from you and toward you. The shadowing is simulated with a tremolo over which you can control the tremolo depth and “width”. The high frequency driver (rotating horn) will have a narrower acoustic beam width (dispersion) than the low frequency driver, and the widths of both may be adjusted. Note that it can take up to one full speaker rotation before you hear changes to tremolo when parameter values are changed. Negative microphone angles take a longer time to respond to tremolo changes than positive microphone angles.

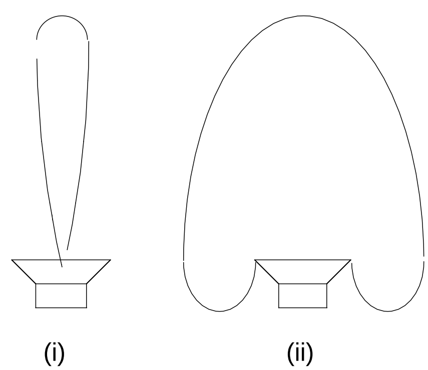


Figure 10-42 Acoustic beams for (i) low frequency driver and (ii) high frequency driver

You can control resonant modes within the rotating speaker cabinet with the Lo and Hi Resonate parameters. For a realistic rotating speaker, the resonance level and delay excursion should be set quite low. High levels will give wild pitch shifting.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Dist Drive	0 to 96 dB
Vib/Chor	V1	DistWarmth	16 to 25088 Hz
Roto InOut	In or Out	Cabinet LP	16 to 25088 Hz

Page 2

Xover	16 to 25088 Hz		
Lo Gain	Off, -79.0 to 24.0 dB	Hi Gain	Off, -79.0 to 24.0 dB
Lo Rate	-10.00 to 10.00 Hz	Hi Rate	-10.00 to 10.00 Hz
Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%
Lo Beam W	45.0 to 360.0 deg	Hi Beam W	45.0 to 360.0 deg

Page 3

LoMicA Pos	-180.0 to 180.0 deg	LoMicB Pos	-180.0 to 180.0 deg
LoMicA Lvl	0 to 100%	LoMicB Lvl	0 to 100%
LoMicA Pan	-100 to 100%	LoMicB Pan	-100 to 100%
HiMicA Pos	-180.0 to 180.0 deg	HiMicB Pos	-180.0 to 180.0 deg
HiMicA Lvl	0 to 100%	HiMicB Lvl	0 to 100%
HiMicA Pan	-100 to 100%	HiMicB Pan	-100 to 100%

Page 4

LoResonate	0 to 100%	HiResonate	0 to 100%
Lo Res Dly	10 to 2550 samp	Hi Res Dly	10 to 2550 samp
LoResXcurs	0 to 510 samp	HiResXcurs	0 to 510 samp
ResH/LPhase	0.0 to 360.0 deg		

In/Out When set to “In”, the algorithm is active; when set to “Off” the algorithm is bypassed.

Out Gain The overall gain or amplitude at the output of the effect. For distortion, it is often necessary to turn the output gain down as the distortion drive is turned up.

VibChInOut When set to “In” the vibrato/chorus is active; when set to “Out” the vibrato/chorus is bypassed.

Vib/Chor This control sets the Hammond B3® vibrato/chorus. There are six settings for this effect: three vibratos “V1”, “V2”, “V3”, and three choruses “C1”, “C2”, “C3”

Roto InOut When set to “In” the rotary speaker is active; when set to “Out” the rotary speaker is bypassed.

Dist Drive Applies a boost to the input signal to overdrive the distortion algorithm. When overdriven, the distortion algorithm will soft-clip the signal. Since distortion drive will make your signal very loud, you may have to reduce the Out Gain as the drive is increased. [VibChor+Rotor 4 only]

DistWarmth A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal. [VibChor+Rotor 4 only]

Cabinet LP A lowpass filter to simulate the band-limiting of a speaker cabinet. The filter controls the upper frequency limit of the output.

Xover The frequency at which high and low frequency bands are split and sent to separate rotating drivers.

Lo Gain The gain or amplitude of the signal passing through the rotating woofer (low frequency driver.

Lo Rate The rotation rate of the rotating woofer (low frequency driver). The woofer can rotate clockwise or counter-clockwise. The direction of rotation depends on the sign of the rate parameter. Assuming microphone angles are set toward the front (between -90° and 90°) and microphones at positive angles are panned to the right (positive pan values), then positive rates correspond to clockwise rotation when viewed from the top.

Lo Size The effective size (radius of rotation) of the rotating woofer in millimeters. Affects the amount of Doppler shift or vibrato of the low frequency signal.

Lo Trem Controls the depth of tremolo of the low frequency signal. Expressed as a percentage of full scale tremolo.

Lo Beam W The rotating speaker effect attempts to model a rotating woofer for the low frequency driver. The acoustic radiation pattern of a woofer tends to range from omnidirectional (radiates in directions in equal amounts) to a wide beam. You may adjust the beam width from 45° to 360°. If you imagine looking down on the rotating speaker, the beam angle is the angle between the -6 dB levels of the beam. At 360°, the woofer is omnidirectional.

Hi Gain The gain or amplitude of the signal passing through the rotating tweeter (high frequency driver.

Hi Rate The rotation rate of the rotating tweeter (high frequency driver). The tweeter can rotate clockwise or counter-clockwise. The direction of rotation depends on the sign of the rate parameter. Assuming microphone angles are set toward the front (between -90° and 90°) and microphones at positive angles are panned to the right (positive pan values), then positive rates correspond to clockwise rotation when viewed from the top.

Hi Size The effective size (radius of rotation) of the rotating tweeter in millimeters. Affects the amount of Doppler shift or vibrato of the high frequency signal.

Hi Trem Controls the depth of tremolo of the high frequency signal. Expressed as a percentage of full scale tremolo.

Hi Beam W The rotating speaker effect attempts to model a rotating horn for the high frequency driver. The acoustic radiation pattern of a horn tends to be a narrow beam. You may adjust the beam width from 45° to 360°. If you imagine looking down on the rotating speaker, the beam angle is the angle between the -6 dB levels of the beam. At 360°, the horn is omnidirectional (radiates in all directions equally).

Mic Pos The angle of the virtual microphones in degrees from the “front” of the rotating speaker. This parameter is not well suited to modulation because adjustments to it will result in large sample skips (audible as clicks when signal is passing through the effect). There are four of these parameters to include 2 pairs (A and B) for high and low frequency drivers.

Mic Lvl The level of the virtual microphone signal being sent to the output. There are four of these parameters to include 2 pairs (A and B) for high and low frequency drivers.

Mic Pan Left-right panning of the virtual microphone signals. A settings of -100% is panned fully left, and 100% is panned fully right. There are four of these parameters to include 2 pairs (A and B) for high and low frequency drivers.

LoResonate A simulation of cabinet resonant modes express as a percentage. For realism, you should use very low settings. This is for the low frequency signal path.

Lo Res Dly The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the low frequency signal path.

LoResXcurs The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the low frequency signal path.

HiResonate A simulation of cabinet resonant modes express as a percentage. For realism, you should use very low settings. This is for the high frequency signal path.

Hi Res Dly The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the high frequency signal path.

HiResXcurs The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the high frequency signal path.

ResH/LPhs This parameter sets the relative phases of the high and low resonators. The angle value in degrees is somewhat arbitrary and you can expect the effect of this parameter to be rather subtle.

734 Distort + Rotary

Small distortion followed by rotary speaker effect

PAUs: 2

Distort + Rotary models an amplifier distortion followed by a rotating speaker. The rotating speaker has separately controllable tweeter and woofer drivers. The algorithm has three main sections. First, the input stereo signal is summed to mono and may be distorted by a tube amplifier simulation. The signal is then passed into the rotator section where it is split into high and low frequency bands and the two bands are run through separate rotators. The two bands are recombined and measured at two positions, spaced by a controllable relative angle (microphone simulation) to obtain a stereo signal again. Finally the signal is passed through a speaker cabinet simulation.

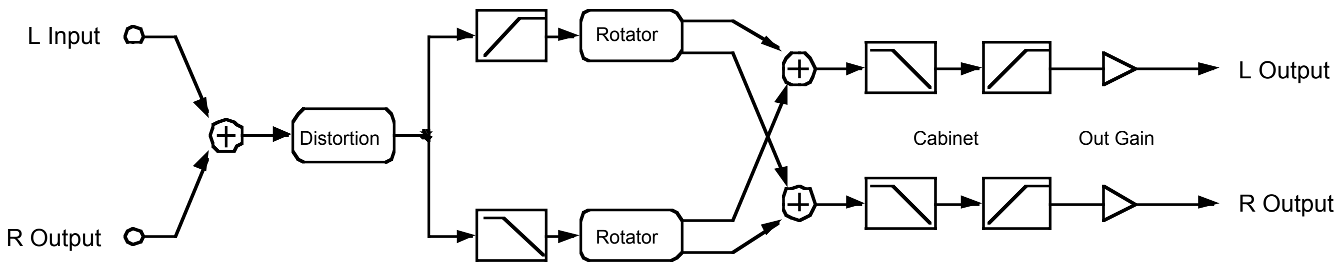


Figure 10-43 Block diagram of Distort + Rotary

The first part of Distort + Rotary is a distortion algorithm. The distortion algorithm will soft clip the input signal. The amount of soft clipping depends on how high the distortion drive parameter is set. Soft clipping means that there is a smooth transition from linear gain to saturated overdrive. Higher distortion drive settings cause the transition to become progressively sharper or “harder”. The distortion never produces hard or digital clipping, but it does approach it at high drive settings. When you increase the distortion drive parameter you are increasing the gain of the algorithm until the signal reaches saturation. You will have to compensate for increases in drive gain by reducing the output gain. These algorithm will not digitally clip unless the output gain is over-driven.

Next the signal passes through a rotating speaker routine. The rotating speaker has separately controllable tweeter and woofer drivers. The signal is split into high and low frequency bands and the two bands are run through separate rotators. The upper and lower rotors each have a pair of virtual microphones which can be positioned at varying positions (angles) around the rotors. The positions of the microphones for the upper and lower drivers is the same. The Mic Angle parameter sets the anglular position of the microphones relative to the loosely defined “front” of the speaker. There are microphones for left and right outputs. As the Mic Angle is increased from 0°, the left microphone moves further to the left and the right microphone moves further to the right. The signal finally passes through a final lowpass and highpass filter pair to simulate the band-limiting effect of the speaker cabinet.

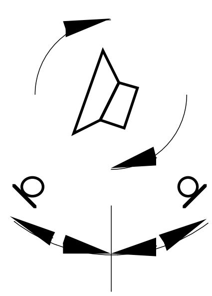


Figure 10-44 Rotating speaker with virtual microphones

For the rotating speakers, you can control the cross-over frequency of the high and low frequency bands (the frequency where the high and low frequencies get separated). The rotating speakers for the high and low frequencies have their own controls. For both, the rotation rate, the effective driver size and tremolo can be set. The rotation rate of course sets how fast the rotating speaker is spinning. The effective driver size is the radius of the path followed by the speaker relative to its center of rotation. This parameter is used to calculate the resulting Doppler shift of the moving speaker. Doppler shift is the pitch shift that occurs when a sound source moves toward or away from you the listener. In a rotating speaker, the Doppler shift will sound like vibrato. As well as Doppler shift, there will be some acoustic shadowing as the speaker is alternately pointed away from you and toward you. The shadowing is simulated with a tremolo over which you can control the tremolo depth.

You can control resonant modes within the rotating speaker cabinet with the Lo and Hi Resonate parameters. For a realistic rotating speaker, the resonance level and delay excursion should be set quite low. High levels will give wild pitch shifting.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Cabinet HP	16 to 25088 Hz	Dist Drive	0 to 96 dB
Cabinet LP	16 to 25088 Hz	DistWarmth	16 to 25088 Hz

Page 2

Xover	16 to 25088 Hz	Mic Angle	0.0 to 360.0 deg
Lo Gain	Off, -79.0 to 24.0 dB	Hi Gain	Off, -79.0 to 24.0 dB
Lo Rate	-10.00 to 10.00 Hz	Hi Rate	-10.00 to 10.00 Hz
Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%

Page 3

		ResH/LPhs	0.0 to 360.0 deg
LoResonate	0 to 100%	HiResonate	0 to 100%
Lo Res Dly	10 to 2550 samp	Hi Res Dly	10 to 2550 samp
LoResXcurs	0 to 510 samp	HiResXcurs	0 to 510 samp

In/Out When set to “In”, the algorithm is active; when set to “Off” the algorithm is bypassed.

Out Gain The overall gain or amplitude at the output of the effect. For distortion, it is often necessary to turn the output gain down as the distortion drive is turned up.

Dist Drive Applies a boost to the input signal to overdrive the distortion algorithm. When overdriven, the distortion algorithm will soft-clip the signal. Since distortion drive will make your signal very loud, you may have to reduce the Out Gain as the drive is increased. [VibChor+Rotor 4 only]

DistWarmth A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal. [VibChor+Rotor 4 only]

Cabinet HP A highpass filter to simulate the band-limiting of a speaker cabinet. The filter controls the lower frequency limit of the output.

Cabinet LP A lowpass filter to simulate the band-limiting of a speaker cabinet. The filter controls the upper frequency limit of the output.

Xover The frequency at which high and low frequency bands are split and sent to separate rotating drivers.

Lo Gain The gain or amplitude of the signal passing through the rotating woofer (low frequency driver.

Lo Rate The rotation rate of the rotating woofer (low frequency driver). The woofer can rotate clockwise or counter-clockwise. The direction of rotation depends on the sign of the rate parameter. Assuming microphone angles are set toward the front (between -90° and 90°) and microphones at positive angles are panned to the right (positive pan values), then positive rates correspond to clockwise rotation when viewed from the top.

Lo Size The effective size (radius of rotation) of the rotating woofer in millimeters. Affects the amount of Doppler shift or vibrato of the low frequency signal.

Lo Trem Controls the depth of tremolo of the low frequency signal. Expressed as a percentage of full scale tremolo.

Hi Gain The gain or amplitude of the signal passing through the rotating tweeter (high frequency driver.

Hi Rate The rotation rate of the rotating tweeter (high frequency driver). The tweeter can rotate clockwise or counter-clockwise. The direction of rotation depends on the sign of the rate parameter. Assuming microphone angles are set toward the front (between -90° and 90°) and microphones at positive angles are panned to the right (positive pan values), then positive rates correspond to clockwise rotation when viewed from the top.

Hi Size The effective size (radius of rotation) of the rotating tweeter in millimeters. Affects the amount of Doppler shift or vibrato of the high frequency signal.

Hi Trem Controls the depth of tremolo of the high frequency signal. Expressed as a percentage of full scale tremolo.

Mic Angle The angle of the virtual microphones in degrees from the "front" of the rotating speaker. For the left microphone the angle increases clockwise (when viewed from the top), while for the right microphone the angle increases counter-clockwise. This parameter is not well suited to modulation because adjustments to it will result in large sample skips (audible as clicks when signal is passing through the effect).

LoResonate A simulation of cabinet resonant modes express as a percentage. For realism, you should use very low settings. This is for the low frequency signal path.

Lo Res Dly The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the low frequency signal path.

LoResXcurs The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the low frequency signal path.

HiResonate A simulation of cabinet resonant modes express as a percentage. For realism, you should use very low settings. This is for the high frequency signal path.

Hi Res Dly The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the high frequency signal path.

HiResXcurs The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the high frequency signal path.

ResH/LPhs This parameter sets the relative phases of the high and low resonators. The angle value in degrees is somewhat arbitrary and you can expect the effect of this parameter to be rather subtle.

KB3 FX Algorithms

735 KB3 FXBus
736 KB3 AuxFX

Vibrato/chorus into distortion into rotating speaker into cabinet

PAUs: 7 for full working effect
4 for KB3 FXBus
3 for KB3 AuxFX

The KB3 FXBus and KB3 AuxFX algorithms contain multiple effects designed for the Hammond B3® emulation (KB3 mode). For correct operation both effects must be running at the same time with the output of KB3 FXBus feeding the input of KB3 AuxFX. The two algorithms work as one algorithm which use all the available KDFX resources. While the input to KB3 FXBus is stereo (which gets summed to mono) and the output from KB3 AuxFX is stereo, the signals between the two algorithms are the low frequency (left) and high frequency (right) signal bands used to drive the lower and upper rotary speakers. It is possible to run these two algorithms as independent effects, but the results will be somewhat unusual, and therefore not generally recommended.

These effects are the Hammond vibrato/chorus, amplifier distortion, and rotating speaker (Leslie®) emulations. Each of these effects may be turned off or bypassed, or the entire algorithm may be bypassed. To bypass the rotary, the switches in both KB3 FXBus and KB3 AuxFX must be set to **Out**.

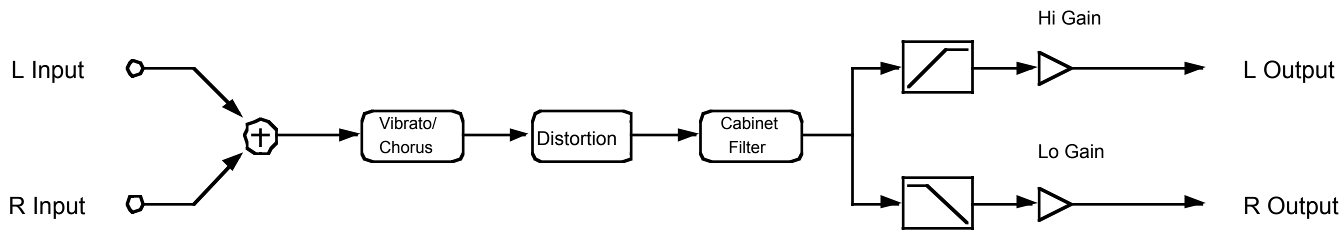


Figure 10-45 Block diagram of KB3 FXBus

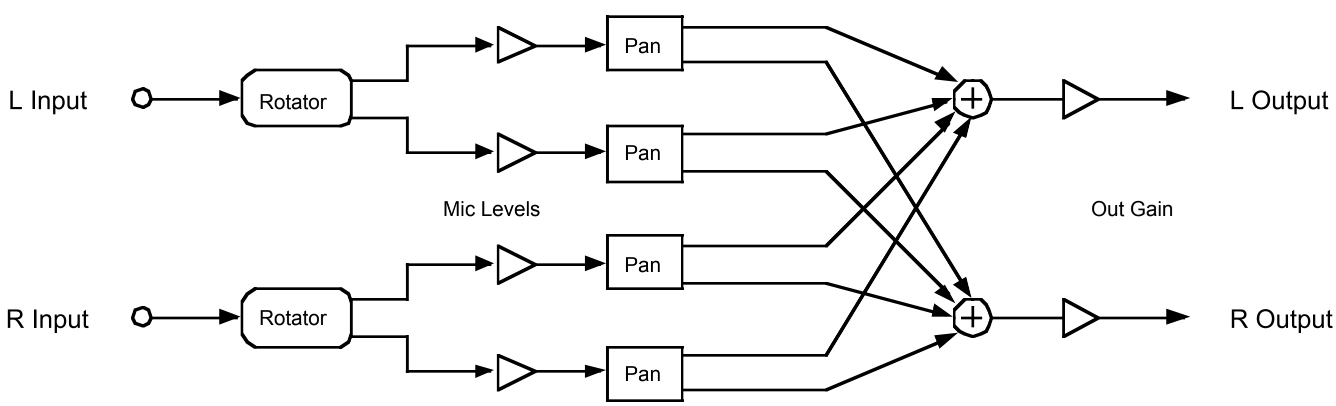


Figure 10-46 Block diagram of KB3 AuxFX

The first effect in the chain is the Hammond vibrato/chorus algorithm. The vibrato/chorus has six settings which are the same as those used in the Hammond B3®: three vibrato (V1, V2, V3) and three chorus (C1, C2, C3) settings. The vibrato chorus has been carefully modelled after the electro-mechanical vibrato/chorus in the B3.

An amplifier distortion algorithm follows the vibrato/chorus. The distortion algorithm will soft clip the input signal. The amount of soft clipping depends on how high the distortion drive parameter is set. Soft clipping means that there is a smooth transition from linear gain to saturated overdrive. Higher distortion drive settings cause the transition to become progressively sharper or “harder”. The distortion never produces hard or digital clipping, but it does approach it at high drive settings. When you increase the distortion drive parameter you are increasing the gain of the algorithm until the signal reaches saturation. You will have to compensate for increases in drive gain by reducing the output gain. These algorithm will not digitally clip unless the output gain is over-driven.

The distorted signal is next passed to a cabinet emulation filter and a pair of crossover filters for band splitting. The measurements of a real Leslie® speaker was used in the design of these filters. Default parameter values reflect these measurements, but you may alter them if you like. The Lo HP parameter controls a highpass filter which defines the lowest frequency to pass through the speaker. Likewise the Hi LP parameter is a lowpass filter controlling the the highest frequency. The crossover filters for the lower and upper drivers may be set independently. A small amount of overlap seems to work well. The gains of the high and low band signals may also be separately controlled.

At this point KB3 FXBus has finished its processing and passes the high and low signals to the KB3 AuxFX algorithm which contains the rotating speaker routine. The rotating speaker has separately controllable tweeter and woofer drivers. The signal is split into high and low frequency bands and the two bands are run through separate rotators. The upper and lower rotors each have a pair of virtual microphones which can be positioned at varying positions (angles) around the rotors. An angle of 0° is loosely defined as the front. You can also control the levels and left-right panning of each virtual microphone. The signal is then passed through a final lowpass filter to simulate the band-limiting effect of the speaker cabinet.

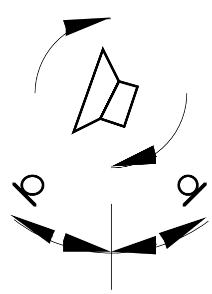


Figure 10-47 Rotating speaker with virtual microphones

The rotating speakers for the high and low frequencies have their own controls. For both, the rotation rate, the effective driver size and tremolo can be set. The rotation rate of course sets how fast the rotating speaker is spinning. The effective driver size is the radius of the path followed by the speaker relative to its center of rotation. This parameter is used to calculate the resulting Doppler shift of the moving speaker. Doppler shift is the pitch shift that occurs when a sound source moves toward or away from you the listener. In a rotating speaker, the Doppler shift will sound like vibrato. As well as Doppler shift, there will be some acoustic shadowing as the speaker is alternately pointed away from you and toward you. The shadowing is simulated with a tremolo over which you can control the tremolo depth and “width”. The high frequency driver (rotating horn) will have a narrower acoustic beam width (dispersion) than the low frequency driver, and the widths of both may be adjusted. Note that it can take up to one full speaker rotation before you hear changes to tremolo when parameter values are changed. Negative microphone angles take a longer time to respond to tremolo changes than positive microphone angles.

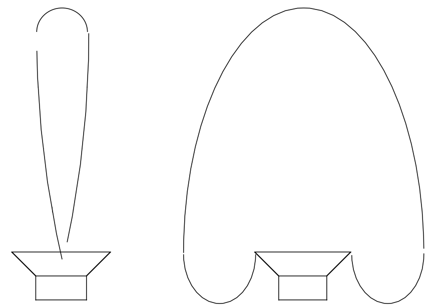


Figure 10-48 Acoustic beams for (i) low frequency driver and (ii) high frequency driver

You can control resonant modes within the rotating speaker cabinet with the Lo and Hi Resonate parameters. For a realistic rotating speaker, the resonance level and delay excursion should be set quite low. High levels will give wild pitch shifting.

Parameters for KB3 FXBus

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Dist Drive	0 to 96 dB
Vib/Chor	V1	DistWarmth	16 to 25088 Hz

Page 2

RotoInOut	In or Out		
Lo Gain	Off, -79.0 to 24.0 dB	Hi Gain	Off, -79.0 to 24.0 dB
Lo Xover	16 to 25088 Hz	Hi Xover	16 to 25088 Hz
Lo HP	16 to 25088 Hz	Hi LP	16 to 25088 Hz

In/Out When set to “In”, the algorithm is active; when set to “Off” the algorithm is bypassed. For the entire algorithm to be active, KB3 AuxFX must also be active.

Out Gain The overall gain or amplitude at the output of the effect. For distortion, it is often necessary to turn the output gain down as the distortion drive is turned up.

VibChInOut When set to “In” the vibrato/chorus is active; when set to “Out” the vibrato/chorus is bypassed.

Vib/Chor This control sets the Hammond B3 vibrato/chorus. There are six settings for this effect: three vibratos “V1”, “V2”, “V3”, and three choruses “C1”, “C2”, “C3”

Roto InOut When set to “In” the rotary speaker is active; when set to “Out” the rotary speaker is bypassed. By bypassing the rotary effect in KB3 FXBus, only the crossover filters are bypassed. You must also bypass KB3 AuxFX to completely bypass the rotary speakers. Likewise, for the entire rotary to be active, KB3 AuxFX must also be active.

Dist Drive Applies a boost to the input signal to overdrive the distortion algorithm. When overdriven, the distortion algorithm will soft-clip the signal. Since distortion drive will make your signal very loud, you may have to reduce the Out Gain as the drive is increased.

Warmth A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal.

Lo Gain The gain or amplitude of the signal passing through the rotating woofer (low frequency driver. The control is also available in KB3 AuxFX.

Lo Xover The crossover frequency for the low frequency driver. Lo Xover controls a lowpass filter.

Lo HP A highpass filter to simulate the band-limiting of a speaker cabinet. The filter controls the lower frequency limit of the output.

Hi Gain The gain or amplitude of the signal passing through the rotating tweeter (high frequency driver. The control is also available in KB3 AuxFX.

Hi Xover The crossover frequency for the high frequency driver. Hi Xover controls a highpass filter.

Hi LP A lowpass filter to simulate the band-limiting of a speaker cabinet. The filter controls the upper frequency limit of the output.

Parameters for KB3 AuxFX

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB

Page 2

Lo Gain	Off, -79.0 to 24.0 dB	Hi Gain	Off, -79.0 to 24.0 dB
Lo Rate	-10.00 to 10.00 Hz	Hi Rate	-10.00 to 10.00 Hz
Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%
Lo Beam W	45.0 to 360.0 deg	Hi Beam W	45.0 to 360.0 deg

Page 3

LoMicA Pos	-180.0 to 180.0 deg	LoMicB Pos	-180.0 to 180.0 deg
LoMicA Lvl	0 to 100%	LoMicB Lvl	0 to 100%
LoMicA Pan	-100 to 100%	LoMicB Pan	-100 to 100%
HiMicA Pos	-180.0 to 180.0 deg	HiMicB Pos	-180.0 to 180.0 deg
HiMicA Lvl	0 to 100%	HiMicB Lvl	0 to 100%
HiMicA Pan	-100 to 100%	HiMicB Pan	-100 to 100%

Page 4

LoResonate	0 to 100%	HiResonate	0 to 100%
Lo Res Dly	10 to 2550 samp	Hi Res Dly	10 to 2550 samp
LoResXcurs	0 to 510 samp	HiResXcurs	0 to 510 samp
ResH/LPhs	0.0 to 360.0 deg		

In/Out When set to “In”, the algorithm is active; when set to “Off” the algorithm is bypassed. For the entire algorithm to be active, KB3 FXBus must also be active with its Roto InOut parameter set to “In”. To completely bypass the rotary, one or both of the In/Out or Roto InOut parameters in KB3 FXBus must also be bypassed.

Out Gain The overall gain or amplitude at the output of the effect.

Lo Gain The gain or amplitude of the signal passing through the rotating woofer (low frequency driver. The control is also available in KB3 FXBus.

Lo Rate The rotation rate of the rotating woofer (low frequency driver). The woofer can rotate clockwise or counter-clockwise. The direction of rotation depends on the sign of the rate parameter. Assuming microphone angles are set toward the front (between -90° and 90°) and microphones at positive angles are panned to the right (positive pan values), then positive rates correspond to clockwise rotation when viewed from the top.

Lo Size The effective size (radius of rotation) of the rotating woofer in millimeters. Affects the amount of Doppler shift or vibrato of the low frequency signal.

Lo Trem Controls the depth of tremolo of the low frequency signal. Expressed as a percentage of full scale tremolo.

Lo Beam W The rotating speaker effect attempts to model a rotating woofer for the low frequency driver. The acoustic radiation pattern of a woofer tends to range from omnidirectional (radiates in directions in equal amounts) to a wide beam. You may adjust the beam width from 45° to 360°. If you imagine looking down on the rotating speaker, the beam angle is the angle between the -6 dB levels of the beam. At 360°, the woofer is omnidirectional.

Hi Gain The gain or amplitude of the signal passing through the rotating tweeter (high frequency driver. The control is also available in KB3 FXBus.

Hi Rate The rotation rate of the rotating tweeter (high frequency driver). The tweeter can rotate clockwise or counter-clockwise. The direction of rotation depends on the sign of the rate parameter. Assuming microphone angles are set toward the front (between -90° and 90°) and microphones at positive angles are panned to the right (positive pan values), then positive rates correspond to clockwise rotation when viewed from the top.

Hi Size The effective size (radius of rotation) of the rotating tweeter in millimeters. Affects the amount of Doppler shift or vibrato of the high frequency signal.

Hi Trem Controls the depth of tremolo of the high frequency signal. Expressed as a percentage of full scale tremolo.

Hi Beam W The rotating speaker effect attempts to model a rotating horn for the high frequency driver. The acoustic radiation pattern of a horn tends to be a narrow beam. You may adjust the beam width from 45° to 360°. If you imagine looking down on the rotating speaker, the beam angle is the angle between the -6 dB levels of the beam. At 360°, the horn is omnidirectional (radiates in all directions equally).

Mic Pos	The angle of the virtual microphones in degrees from the "front" of the rotating speaker. This parameter is not well suited to modulation because adjustments to it will result in large sample skips (audible as clicks when signal is passing through the effect). There are four of these parameters to include 2 pairs (A and B) for high and low frequency drivers.
Mic Lvl	The level of the virtual microphone signal being sent to the output. There are four of these parameters to include 2 pairs (A and B) for high and low frequency drivers.
Mic Pan	Left-right panning of the virtual microphone signals. A settings of -100% is panned fully left, and 100% is panned fully right. There are four of these parameters to include 2 pairs (A and B) for high and low frequency drivers.
LoResonate	A simulation of cabinet resonant modes express as a percentage. For realism, you should use very low settings. This is for the low frequency signal path.
Lo Res Dly	The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the low frequency signal path.
LoResXcurs	The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the low frequency signal path.
HiResonate	A simulation of cabinet resonant modes express as a percentage. For realism, you should use very low settings. This is for the high frequency signal path.
Hi Res Dly	The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the high frequency signal path.
HiResXcurs	The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the high frequency signal path.
ResH/LPhs	This parameter sets the relative phases of the high and low resonators. The angle value in degrees is somewhat arbitrary and you can expect the effect of this parameter to be rather subtle.

900 Env Follow Filt

Envelope following stereo 2 pole resonant filter

PAUs: 2

The envelope following filter is a stereo resonant filter with the resonant frequency controlled by the envelope of the input signal (the maximum of left or right). The filter type is selectable and may be one of low pass (i), high pass (ii), band pass (iii), or notch (iv).

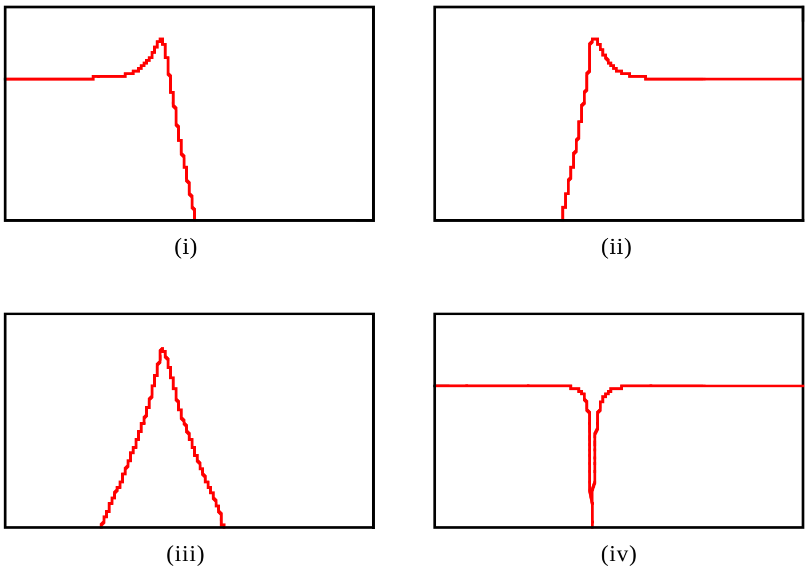


Figure 10-49 Resonant Filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

The resonant frequency of the filter will remain at the minimum frequency (Min Freq) as long as the signal envelope is below the Threshold. The Freq Sweep parameter controls how much the frequency will change with changes in envelope amplitude. The frequency range is 0 to 8372 Hz, though the minimum setting for Min Freq is 58 Hz. Note that the term minimum frequency is a reference to the resonant frequency at the minimum envelope level; with a negative Freq Sweep, the filter frequency will sweep below the Min Freq. A meter is provided to show the current resonance frequency of the filter.

The filter Resonance level may be adjusted. The resonance is expressed in decibels (dB) of gain at the resonant frequency. Since 50 dB of gain is available, you will have to be careful with your gain stages to avoid clipping.

The attack and release rates of the envelope follower are adjustable. The rates are expressed in decibels per second (dB/s). The envelope may be smoothed by a low pass filter which can extend the attack and release times of the envelope follower. A level meter with a threshold marker is provided.

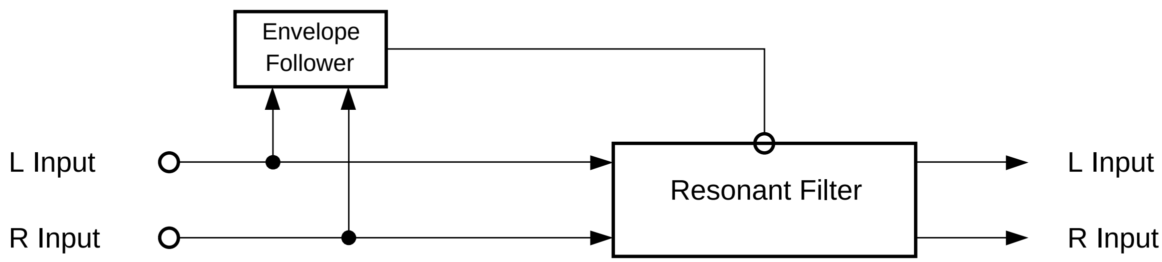


Figure 10-50 Block diagram of envelope following filter

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
FilterType	Lowpass	Min Freq	58 to 8372 Hz
F		Freq Sweep	-100 to 100%
0Hz 2k 4k 6k		Resonance	0 to 50 dB

Page 2

Threshold	-79.0 to 0.0 dB	Atk Rate	0.0 to 300.0 dB/s
		Rel Rate	0.0 to 300.0 dB/s
		Smth Rate	0.0 to 300.0 dB/s

Wet/Dry The amount of modulated (wet) signal relative to unaffected (dry) signal as a percent.

Out Gain The overall gain or amplitude at the output of the effect.

FilterType The type of resonant filter to be used. Lowpass, Highpass, Bandpass, or Notch.

Min Freq The base frequency of the resonant filter. The filter resonant frequency is set to the Min Freq while the signal envelope is at its minimum level or below the threshold.

Freq Sweep How far the filter frequency can change from the Min Freq setting as the envelope amplitude changes. Freq Sweep may be positive or negative so the filter frequency can rise above or fall below the Min Freq setting.

Resonance The resonance level of the resonant filter. Resonance sets the level of the resonant peak (or the amount of cut in the case of the notch filter).

Threshold Tthe level above which signal envelope must rise before the filter begins to follow the envelope. Below the threshold, the filter resonant frequency remains at the Min frequency.

Atk Rate Adjusts the upward slew rate of the envelope detector.

Rel Rate Adjusts the downward slew rate of the envelope detector.

Smth Rate Smooths the output of the envelope follower. Smoothing slows down the envelope follower and can dominate the attack and release rates if set to a lower rate than either of these parameters.

901 TrigEnvelopeFilt

Triggered envelope following stereo 2 pole resonant filter

PAUs: 2

The triggered envelope following filter is used to produce a filter sweep when the input rises above a trigger level. The triggered envelope following filter is a stereo resonant filter with the resonant frequency controlled by a triggered envelope follower. The filter type is selectable and may be one of low pass (i), high pass (ii), band pass (iii), or notch (iv).

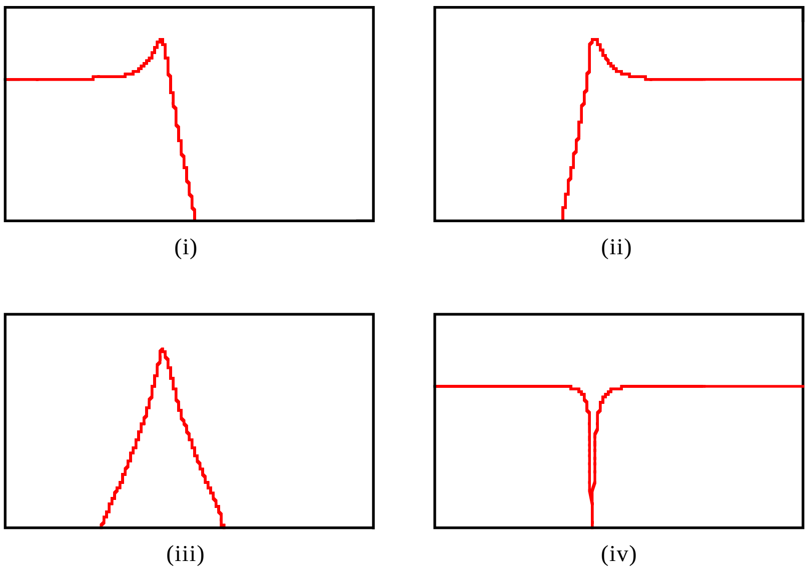


Figure 10-51 Resonant Filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

The resonant frequency of the filter will remain at the minimum frequency (Min Freq) prior to being triggered. On a trigger, the resonant frequency will sweep to the maximum frequency (Max Freq). The minimum and maximum frequencies may be set to any combination of frequencies between 58 and 8372 Hz. Note that the terms minimum and maximum frequency are a reference to the resonant frequencies at the minimum and maximum envelope levels; you may set either of the frequencies to be larger than the other. A meter is provided to show the current resonance frequency of the filter.

The filter Resonance level may be adjusted. The resonance is expressed in decibels (dB) of gain at the resonant frequency. Since 50 dB of gain is available, you will have to be careful with your gain stages to avoid clipping.

When the input signal envelope rises above the trigger level, an envelope generator is started which has an instant attack and exponential decay. The generated attack may be lengthened with the the smoothing parameter. The smoothing parameter can also lengthen the generated decay if the smoothing rate is lower than the decay. The generated envelope is then used to control the resonant frequency of the filter.

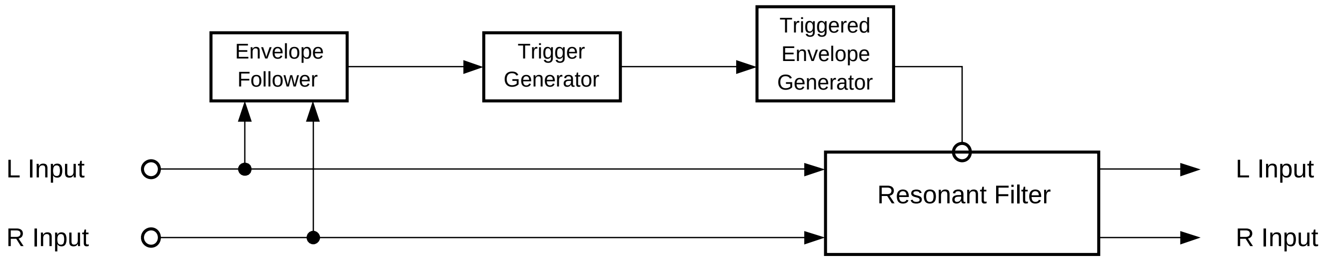


Figure 10-52 Block diagram of Triggered Envelope Filter

The time constant of the envelope follower may be set (Env Rate) as well as the decay rate of the generated envelope (Rel Rate). After the detected envelope rises above the Trigger level, a trigger event cannot occur again until the signal drops below the Retrigger level. In general, Retrigger should be set lower than the Trigger level. A level meter with a trigger marker is provided.

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
FilterType	Lowpass	Min Freq	58 to 8372 Hz
F		Max Freq	58 to 8372 Hz
0Hz 2k 4k 6k		Resonance	0 to 50 dB

Page 2

Trigger	-79.0 to 0.0 dB	Env Rate	0.0 to 300.0 dB/s
Retrigger	-79.0 to 0.0 dB	Rel Rate	0.0 to 300.0 dB/s
		Smth Rate	0.0 to 300.0 dB/s

Wet/Dry The amount of modulated (wet) signal relative to unaffected (dry) signal as a percent.

Out Gain The overall gain or amplitude at the output of the effect.

FilterType The type of resonant filter to be used. May be one of “Lowpass”, “Highpass”, “Bandpass”, or “Notch”.

Min Freq The base frequency of the resonant filter. The filter resonant frequency is set to the base frequency while the signal envelope is below the threshiold.

Max Freq The frequency of the resonant filter that can be reached when the envelope follower output reaches full-scale. The resonant frequency will sweep with the envelope from the base frequency, approaching the limit frequency with rising amplitudes.

Resonance The resonance level of the resonant filter. Resonance sets the level of the resonant peak (or the amount of cut in the case of the notch filter).

Trigger The threshold at which the envelope detector triggers in fractions of full scale where 0dB is full scale.

Retrigger The threshold at which the envelope detector resets such that it can trigger again in fractions of full scale where 0dB is full scale. This value is only useful when it is below the value of Trigger.

Env Rate The envelope detector decay rate which can be used to prevent false triggering. When the signal envelope falls below the retrigger level, the filter can be triggered again when the signal rises above the trigger level. Since the input signal can fluctuate rapidly, it is necessary to adjust the rate at which the signal envelope can fall to the retrigger level. The rate is provided in decibels per second (dB/s).

Rel Rate The downward slew rate of the triggered envelope generator. The rate is provided in decibels per second (dB/s).

Smth Rate Smooths the output of the envelope generator. Smoothing slows down the envelope follower and can dominate the release rate if set lower rate than this parameter. You can use the smoothing rate to lengthen the attack of the generated envelope which would otherwise have an instant attack. The rate is provided in decibels per second (dB/s).

902 LFO Sweep Filter

LFO following stereo 2 pole resonant filter

PAUs: 2

The LFO following filter is a stereo resonant filter with the resonant frequency controlled by an LFO (low-frequency oscillator). The filter type is selectable and may be one of low pass (i), high pass (ii), band pass (iii), or notch (iv) (see figure below).

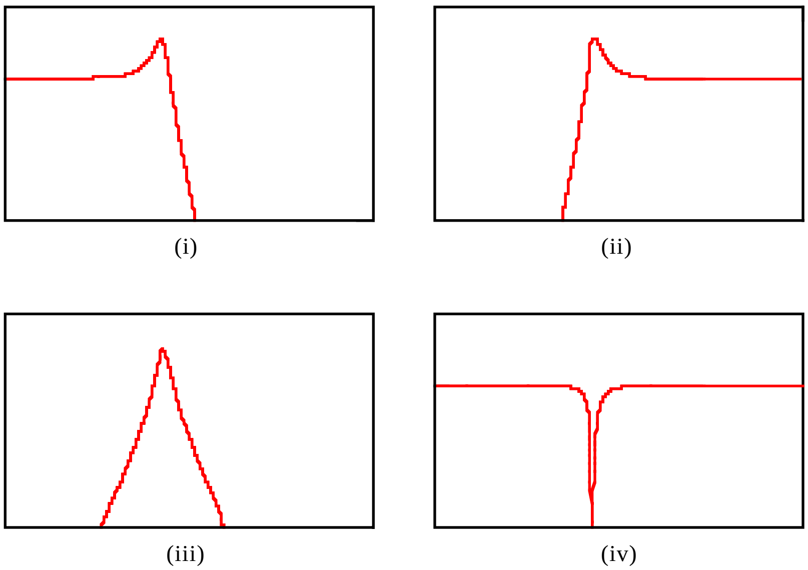


Figure 10-53 Resonant Filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

The resonant frequency of the filter will sweep between the minimum frequency (Min Freq) and the maximum frequency (Max Freq). The minimum and maximum frequencies may be set to any combination of frequencies between 58 and 8372 Hz. Note that the terms minimum and maximum frequency are a reference to the resonant frequencies at the minimum and maximum envelope levels; you may set either of the frequencies to be larger than the other, though doing so will just invert the direction of the LFO. Meters are provided to show the current resonance frequencies of the left and right channel filters.

The filter Resonance level may be adjusted. The resonance is expressed in decibels (dB) of gain at the resonant frequency. Since 50 dB of gain is available, you will have to be careful with your gain stages to avoid clipping.

You can set the frequency of the LFO using the LFO Tempo and LFO Period controls. You can explicitly set the tempo or use the system tempo from the sequencer (or MIDI clock). The LFO Period control sets the period of the LFO (the time for one complete oscillation) in terms of the number of tempo beats per LFO period. The LFO may be configured to one of a variety of wave shapes. Available shapes are Sine, Saw+, Saw-, Pulse and Tri (Figure 2). Sine is simply a sinusoid waveform. Tri produces a triangular waveform, and Pulse produces a series of square pulses where the pulse width can be adjusted with the “LFO PlsWid” parameter. When pulse width is 50%, the signal is a square wave. The “LFO PlsWid” parameter is only active when the Pulse waveform is selected. Saw+ and Saw- produce rising and falling sawtooth waveforms. The Pulse and Saw waveforms have abrupt, discontinuous changes in amplitude which can be smoothed. The pulse wave is implemented as a hard clipped sine wave, and, at 50% width, it turns into a sine wave when set to 100% smoothing. The sudden change in amplitude of the sawtooths develops a more gradual slope with smoothing, ending up as triangle waves when set to 100% smoothing.

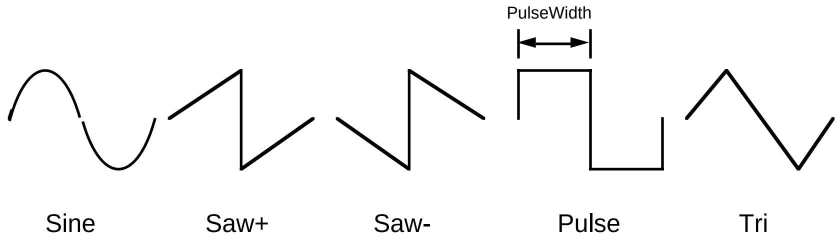


Figure 10-54 Configurable Wave Shapes

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
LFO Tempo	System, 1 to 255 BPM	LFO Shape	Sine
LFO Period	1/24 to 32 bts	LFO PlsWid	0 to 100%
		LFO Smooth	0 to 100%

Page 2

FilterType	Lowpass	Min Freq	58 to 8372 Hz
		Max Freq	58 to 8372 Hz
		Resonance	0 to 50 dB
L Phase	0.0 to 360.0 deg	R Phase	0.0 to 360.0 deg
L		R	
0Hz 2k 4k 6k		0Hz 2k 4k 6k	

Wet/Dry The amount of modulated (wet) signal relative to unaffected (dry) signal as a percent.

Out Gain The overall gain or amplitude at the output of the effect.

LFO Tempo Basis for the rates of the LFO, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LFO Period Sets the LFO rate based on the Tempo determined above: the number of beats corresponding to one period of the LFO cycle. For example, if the LFO Period is set to “4”, the LFOs will take four beats to pass through one oscillation, so the LFO rate will be 1/4th of the Tempo setting. If it is set to “6/24” (=1/4), the LFO will oscillate four times as fast as the Tempo. At “0”, the LFOs stop oscillating and their phase is undetermined (wherever they stopped).

LFO Shape The waveform type for the LFO. Choices are Sine, Saw+, Saw-, Pulse, and Tri.

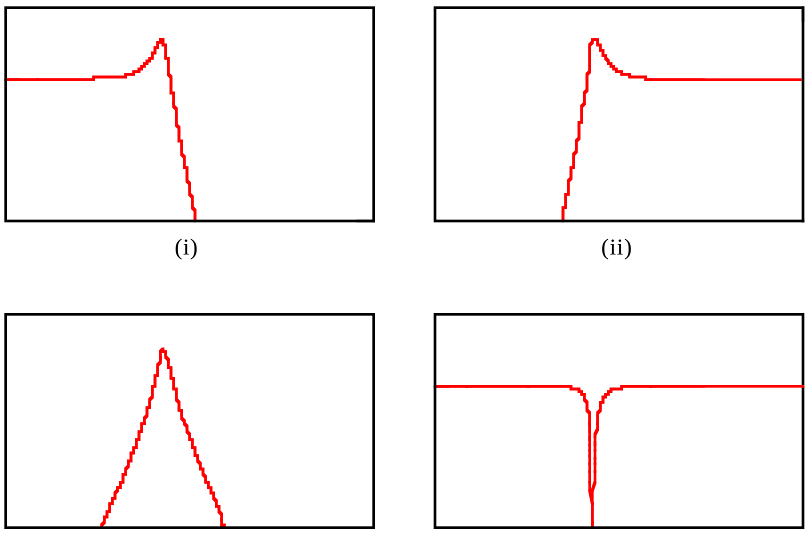
LFO PlsWid	When the LFO Shape is set to Pulse, the PlsWid parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to 50%. This parameter is active only when the Pulse waveform is selected.
LFO Smooth	Smooths the Saw+, Saw-, and Pulse waveforms. For the sawtooth waves, smoothing makes the waveform more like a triangle wave. For the Pulse wave, smoothing makes the waveform more like a sine wave.
FilterType	The type of resonant filter to be used. May be one of “Lowpass”, “Highpass”, “Bandpass”, or “Notch”.
Min Freq	The minimum frequency of the resonant filter. This is the resonant frequency at one of the extremes of the LFO sweep. The resonant filter frequency will sweep between the Min Freq and Max Freq.
Max Freq	The maximum frequency of the resonant filter. This is resonant frequency at the other extreme of the LFO sweep. The resonant filter frequency will sweep between the Min Freq and Max Freq.
Resonance	The resonance level of the resonant filter. Resonance sets the level of the resonant peak (or the amount of cut in the case of the notch filter).
L Phase	The phase angle of the left channel LFO relative to the system tempo clock and the right channel phase.
R Phase	The phase angle of the right channel LFO relative to the system tempo clock and the left channel phase.

903 Resonant Filter
904 Dual Res Filter

Stereo and dual mono 2 pole resonant filters

PAUs: 1 for Resonant Filter
1 for Dual Res Filter

The resonant filter is available as a stereo (linked parameters for left and right) or dual mono (independent controls for left and right). The filter type is selectable and may be one of low pass (i), high pass (ii), band pass (iii), or notch (iv) (see figure below).



(i) (ii)
(iii) (iv)

Figure 10-55 Resonant Filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

You can adjust he resonant frequency of the filter and the filter resonance level.

Parameters for Resonant Filter

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
FilterType	Lowpass		
Frequency	58 to 8372 Hz		
Resonance	0 to 50 dB		

Parameters for Dual Res Filter

Page 1

L Wet/Dry	0 to 100%wet	R Wet/Dry	0 to 100%wet
L Output	Off, -79.0 to 24.0 dB	R Output	Off, -79.0 to 24.0 dB

Page 2

L FiltType	Lowpass	R FiltType	Highpass
L Freq	58 to 8372 Hz	R Freq	58 to 8372 Hz
LResonance	0 to 50 dB	RResonance	0 to 50 dB

Wet/Dry The amount of filtered (wet) signal relative to unaffected (dry) signal.

Out Gain The overall gain or amplitude at the output of the filter.

FilterType The type of resonant filter to be used. May be one of “Lowpass”, “Highpass”, “Bandpass”, or “Notch”.

Frequency The frequency of the resonant filter peak (or notch) in Hz. The frequencies correspond to semitone increments.

Resonance The resonance level of the resonant filter. Resonance sets the level of the resonant peak (or the amount of cut in the case of the notch filter).

905 EQ Morpher/ 906 Mono EQ Morpher

Parallel resonant bandpass filters with parameter morphing

PAUs: 4 for EQ Morpher
2 for Mono EQ Morpher

The EQ Morpher algorithms have four parallel bandpass filters acting on the input signal and the filter results are summed for the final output. EQ Morpher is a stereo algorithm for which the left and right channels receive separate processing using the same linked controls. Mono EQ Morpher sums the input left and right channels into a mono signal, so there is only one channel of processing. Both algorithms have output panning. In EQ Morpher, a stereo panner like that in INPUT page is used and includes a width parameter to control the width of the stereo field. Mono EQ Morph uses a standard mono panner for positioning the mono signal between the left and right speakers.

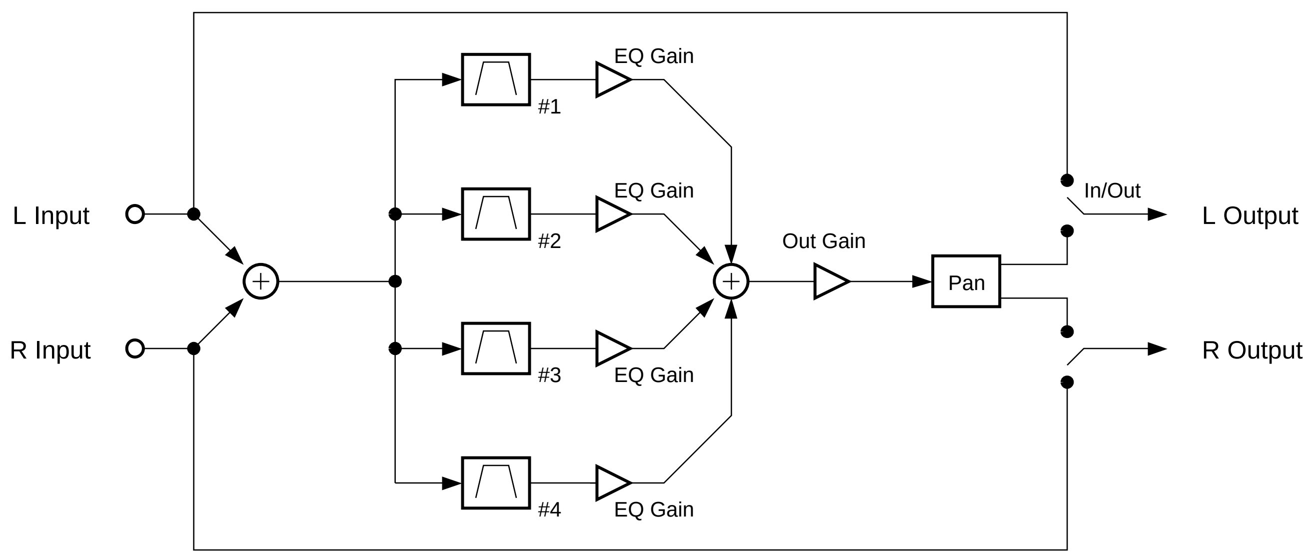


Figure 10-56 Mono EQ Morpher (EQ Morpher is similar)

For each filter, there are two sets of parameters, A and B. The parameter Morph A>B determines which parameter set is active. When Morph A>B is set to 0%, you are hearing the A parameters; when set to 100%, you are hearing the B parameters. The filters may be gradually moved from A to B and back again by moving the Morph A>B parameter between 0 and 100%.

The four filters are parametric bandpass filters. These are not the usual parametric filters you are familiar with. Normal parametric filters boost or cut the signal at the frequency you specify relative to the signal at other frequencies. The bandpass filters used here pass only signals at the frequency you specify and cut all other frequencies. The gain controls for the filters set the levels of each filter's output. Like the normal parametric filters, you have control of the filters' frequencies and bandwidths. The Freq Scale parameters may be used to adjust the A or B filters' frequencies as a group. This allows you to maintain a constant spectral relationship between your filters while adjusting the frequencies up and down. The filters are arranged in parallel and their outputs summed, so the bandpass peaks are added together and the multiple resonances are audible.

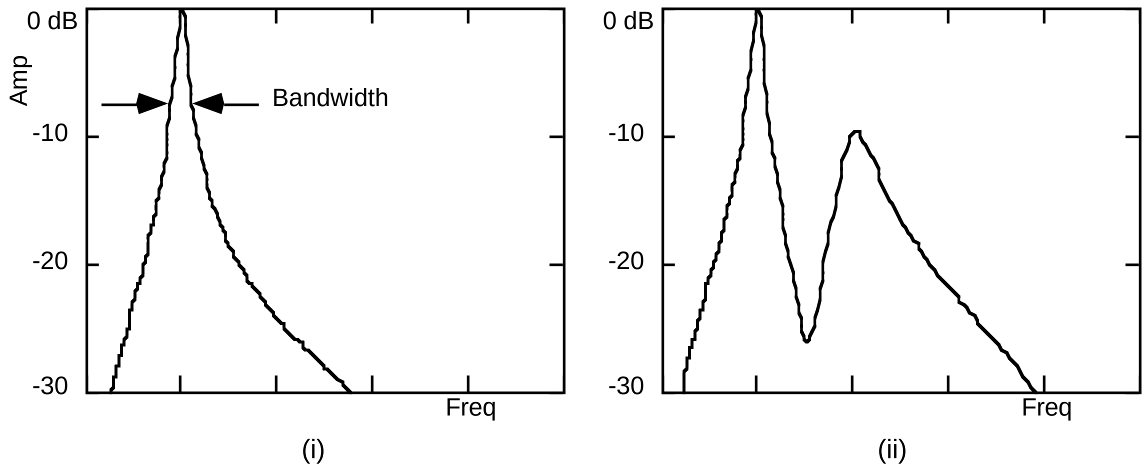


Figure 10-57 Frequency response of (i) a single bandpass filter; (ii) the sum of two bandpass filters

Now that we've gone through what the algorithm does, the question becomes "Why are we doing this?" With careful thought to parameter settings, EQ Morph does an excellent job of simulating the resonances of the vocal tract. A buzz or sawtooth signal is a good choice of source material to experiment with the EQ Morphers. Set the Morph A>B parameter to 0%, and find a combination of A filter settings which give an interesting vowel like sound. It may help to start from existing ROM presets. Next set Morph A>B to 100% and set the B parameters to a different vowel-like sound. You can now set up some FXMods on Morph A>B to morph between the two sets of parameters, perhaps using Freq Scale to make it more expressive.

When morphing from the A parameters to the B parameters, A filter #1 moves to B filter #1, A filter #2 moves to B filter #2, and so on. For the most normal and predictable results, it's a good idea not to let the frequencies of the filters cross each other during the morphing. You can ensure this doesn't happen by making sure the four filters are arranged in ascending order of frequencies. Descending order is okay too, provided you choose an order and stick to it.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Morph A>B	0 to 100%	Out Pan	-100 to 100%
		Out Width[1]	-100 to 100%
AFreqScale	-8600 to 8600 ct	BFreqScale	-8600 to 8600 ct

1. EQ Morpher only

Page 2

A Freq 1	16 to 25088 Hz	B Freq 1	16 to 25088 Hz
A Width 1	0.010 to 5.000 oct	B Width 1	0.010 to 5.000 oct
A Gain 1	-79.0 to 24.0 dB	B Gain 1	-79.0 to 24.0 dB
A Freq 2	16 to 25088 Hz	B Freq 2	16 to 25088 Hz
A Width 2	0.010 to 5.000 oct	B Width 2	0.010 to 5.000 oct
A Gain 2	-79.0 to 24.0 dB	B Gain 2	-79.0 to 24.0 dB

Page 3

A Freq 3	16 to 25088 Hz	B Freq 3	16 to 25088 Hz
A Width 3	0.010 to 5.000 oct	B Width 3	0.010 to 5.000 oct
A Gain 3	-79.0 to 24.0 dB	B Gain 3	-79.0 to 24.0 dB
A Freq 4	16 to 25088 Hz	B Freq 4	16 to 25088 Hz
A Width 4	0.010 to 5.000 oct	B Width 4	0.010 to 5.000 oct
A Gain 4	-79.0 to 24.0 dB	B Gain 4	-79.0 to 24.0 dB

In/Out When set to “In” the algorithm is active; when set to “Out” the algorithm is bypassed.

Out Gain An overall level control of the EQ Morpher output.

Out Pan Provides panning of the output signal between left and right output channels. A setting of -100% is panned left and 100% is panned right. For EQ Morph, this is a stereo panner which pans the entire stereo image as is done with the input sends on the INPUT page when set to the “SP” mode.

Out Width The width of the stereo field is controlled by this parameter. A setting of 100% is the same full width as the input signal. At 0% the left and right channels are narrowed to the point of being mono. Negative values reverse the left and right channels. This parameter is available in EQ Morpher and not Mono EQ Morpher.

Morph A>B When set to 0% the “A” parameters are controlling the filters, and when set to 100%, the “B” parameters control the filters. Between 0 and 100%, the filters are at interpolated positions. When morphing from A to B settings, the A filter #1 will change to the B filter #1, A filter #2 moves to B filter #2, and so on.

FreqScale The filter frequencies for the A and B parameter sets may be offset with the FreqScale parameters. After setting the filter parameters, the FreqScale parameters will move each of the four filter frequencies together by the same relative pitch.

For the two filter sets A & B, there are four filters 1, 2, 3 and 4:

Freq The center frequency of the bandpass filter peak in Hz. This frequency may be offset by the FreqScale parameter.

Width The bandwidth of the bandpass filter in octaves. Narrow bandwidths provide the most convincing vocal sounds.

Gain The level of the bandpass filter output. At 0 dB, a sine wave at the same frequency as the filter will be neither boost not cut. At settings greater than 0 dB, the (hypothetical) sine wave is boosted, and below 0 dB the sine wave is cut. Signals at frequencies other than the filter frequency are always cut more than a signal at the filter frequency. The amount that other frequencies are cut depends on the bandwidth of the bandpass filter.

907 Ring Modulator

A configurable ring modulator

PAUs: 1

Ring modulation is a simple effect in which two signals are multiplied together. Typically, an input signal is modulated with a simple carrier waveform such as a sine wave or a sawtooth. Since the modulation is symmetric (a*b = b*a), deciding which signal is the carrier and which is the modulation signal is a question of perspective. A simple, unchanging waveform is generally considered the carrier.

To see how the ring modulator works, we'll have to go through a little high school math and trigonometry. If you like, you can skip the how's and why's and go straight to the discussion of controlling the algorithm. Let's look at the simple case of two equal amplitude sine waves modulating each other. Real signals will be more complex, but they will be much more difficult to analyse. The two sine waves generally will be oscillating at different frequencies. A sine wave signal at any time t having a frequency f is represented as $\sin(ft + \phi)$ where ϕ is constant phase angle to correct for the sine wave not being 0 at $t = 0$. The sine wave could also be represented with a cosine function which is a sine function with a 90° phase shift. To simply matters, we will write $A = f_1 t + \phi_1$ for one of the sine waves and $B = f_2 t + \phi_2$ for the other sine wave. The ring modulator multiplies the two signals to produce sin A sin B. We can try to find a trigometric identity for this, or we can just look up in a trigonometry book:

$2 \sin A \sin B = \cos(A - B) - \cos(A + B)$.

This equation tells us that multiplying two sine waves produces two new sine waves (or cosine waves) at the sum and difference of the original frequencies. The following figure shows the output frequencies (solid lines) for a given input signal pair (dashed lines):

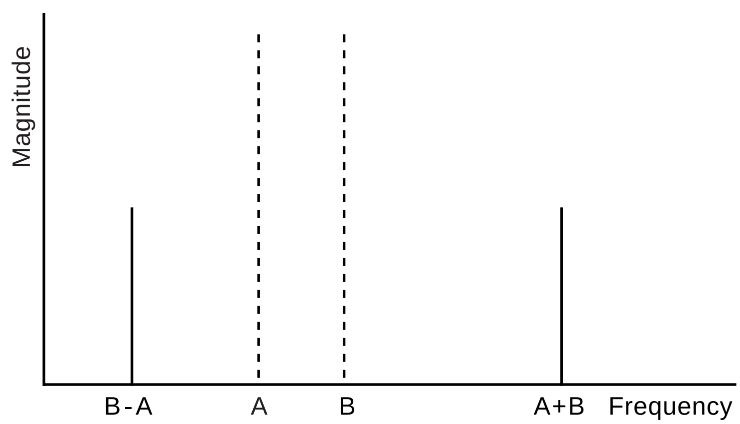


Figure 10-58 Result of Modulating Two Sine Waves A and B

This algorithm has two operating modes which is set with the Mod Mode parameter. In "L*R" mode, you supply the modulation and carrier signals as two mono signals on the left and right inputs. The output in "L*R" mode is also mono and you may use the L*R Pan parameter to pan the output. The oscillator parameters on parameter pages 2 and three will be inactive while in “L*R” mode. Figure 2 shows the signal flow when in “L*R” mode:

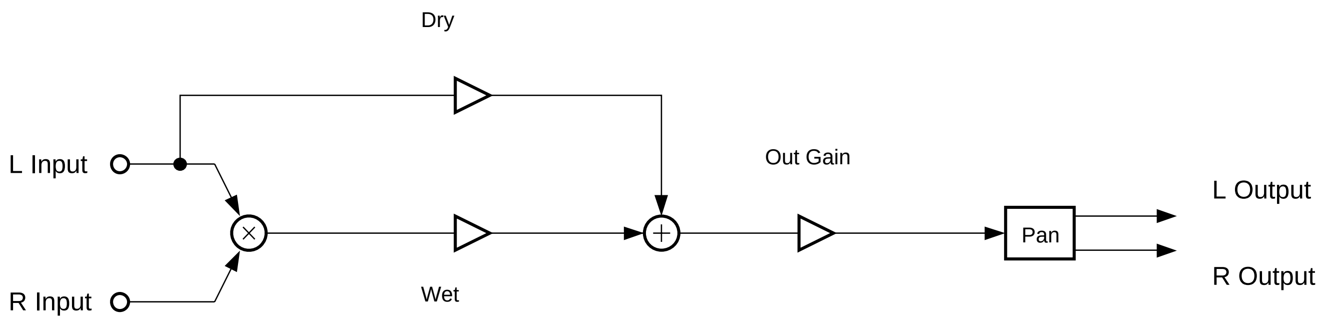


Figure 10-59 “L*R” Mode Ring Modulator

The other modulation mode is “Osc”. In “Osc” mode, the algorithm inputs and outputs are stereo, and the carrier signal for both channels is generated inside the algorithm. The carrier signal is the sum of five oscillators (see Figure 10-60).

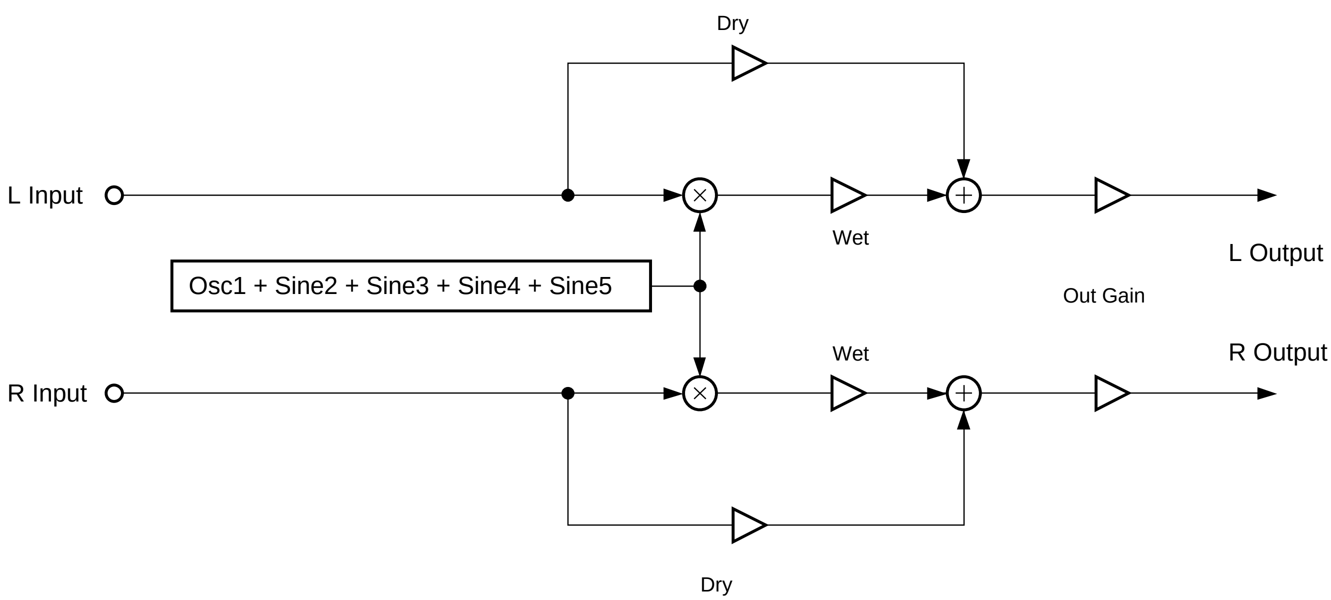


Figure 10-60 “Osc” Mode Ring Modulator

Four of the oscillators are simple sine waves and a fifth may be configured to one of a variety of wave shapes. With all oscillators, you can set level and frequency. The configurable oscillator also lets you set the wave shape. Available shapes are Sine, Saw+, Saw-, Pulse, Tri and Expon (Figure 4). Sine is simply another sine waveform. Tri produces a triangular waveform, and Expon produces a waveform with narrow, sharp peaks which seems to rise exponentially from 0. Pulse produces a series of square pulses where the pulse width can be adjusted with the “Osc1PlsWid” parameter. When pulse width is 50%, the signal is a square wave. The “Osc1PlsWid” parameter is only active when the Pulse waveform is selected. Saw+ and Saw- produce rising and falling sawtooth waveforms. The Pulse and Saw waveforms have abrupt, discontinuous changes in amplitude which can be smoothed. The pulse wave is implemented as a hard clipped sine wave, and, at 50% width, it turns into a sine wave when set to 100% smoothing. The sudden change in amplitude of the sawtooths develops a more gradual slope with smoothing, ending up as triangle waves when set to 100% smoothing.

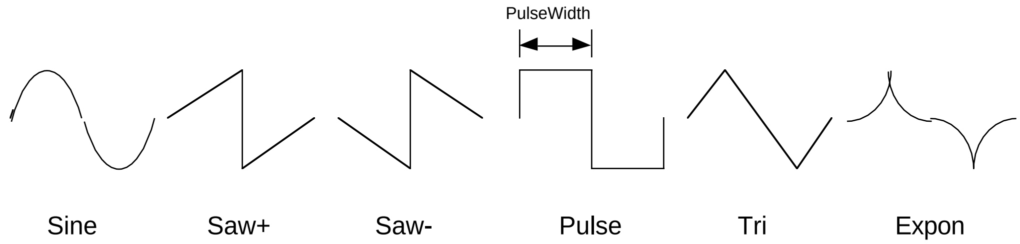


Figure 10-61 Configurable Wave Shapes

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Mod Mode	L*R or Osc	L*R Gain	Off, -79.0 to 48.0 dB
		L*R Pan	-100 to 100%

Page 2

Osc1 Lvl	0 to 100%	Osc1 Freq	16 to 25088 Hz
Osc1 Shape	Sine		
Osc1PlsWid	0 to 100%		
Osc1Smooth	0 to 100%		

Page 3

Sine2 Lvl	0 to 100%	Sine2 Freq	16 to 25088 Hz
Sine3 Lvl	0 to 100%	Sine3 Freq	16 to 25088 Hz
Sine4 Lvl	0 to 100%	Sine4 Freq	16 to 25088 Hz
Sine5 Lvl	0 to 100%	Sine5 Freq	16 to 25088 Hz

Wet/Dry The amount of modulated (wet) signal relative to unaffected (dry) signal as a percent. When in “L*R” mode, the left input will be used as the dry signal.

Out Gain The overall gain or amplitude at the output of the effect.

Mod Mode Switches between the two operating modes of the algorithm. The “L*R” mode treats the left and right inputs as the modulator and carrier signals. It does not matter which input is left and which is right except to note that only the left signal will be passed through as dry.

L*R Pan The output panning of the both wet and dry signals. This control is active only in “L*R” mode. -100% is panned fully left, 0% is panned center and 100% is panned right.

Osc1 Lvl The level of the configurable oscillator. 0% is off and 100% is maximum. This parameter is active only in “Osc” mode.

Osc1 Freq	The fundamental frequency of the configurable oscillator. The oscillators can be set through the audible frequencies 16-25088 Hz with 1 semitone resolution. This parameter is active only in “Osc” mode.
Osc1Shape	Shape selects the waveform type for the configurable oscillator. Choices are Sine, Saw+, Saw-, Pulse, Tri, and Expon. This parameter is active only in “Osc” mode.
Osc1PlsWid	When the configurable oscillator is set to Pulse, the PlsWid parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to 50%. This parameter is active only in “Osc” mode and when the Pulse waveform is selected.
Osc1Smooth	Smooths the Saw+, Saw-, and Pulse waveforms. For the sawtooth waves, smoothing makes the waveform more like a triangle wave. For the Pulse wave, smoothing makes the waveform more like a sine wave.
Sine*n* Lvl	The four sine wave oscillators (*n* = 2...5) may have their levels set between 0% (off) and 100% (maximum). This parameter is active only in “Osc” mode.
Sine*n* Freq	The four sine wave oscillators (*n* = 2...5) may have their frequencies set with this parameter. The oscillators can be set through the audible frequencies 16-25088 Hz with 1 semitone resolution. This parameter is active only in “Osc” mode.

908 Pitcher

Creates pitch from pitched or non-pitched signal

PAUs: 1

This algorithm applies a filter which has a series of peaks in the frequency response to the input signal. The peaks may be adjusted so that their frequencies are all multiples of a selectable frequency, all the way up to 24 kHz. When applied to a sound with a noise-like spectrum (white noise, with a flat spectrum, or cymbals, with a very dense spectrum of many individual components), an output is produced which sounds very pitched, since most of its spectral energy ends up concentrated around multiples of a fundamental frequency.

If the original signal has no signficant components at the desired pitch or harmonics, the output level remains low. The left and right inputs are processed independently with common controls of pitch and weighting. Applying Pitcher to sounds such as a single sawtooth wave will tend to not produce much output, unless the sawtooth frequency and the Pitcher frequency match or are harmonically related, because otherwise the peaks in the input spectrum won't line up with the peaks in the Pitcher filter. If there are enough peaks in the input spectrum (obtained by using sounds with noise components, or combining lots of different simple sounds, especially low pitched ones, or severly distorting a simple sound) then Pitcher can do a good job of imposing its pitch on the sound.

The four weight parameters named “Odd Wts”, “Pair Wts”, “Quartr Wts” and “Half Wts” control the exact shape of the frequency response of Pitcher. An exact description of what each one does is, unfortunately, impossible, since there is a great deal of interaction between them. Here are some examples with a Pitch setting of 1 Khz, which is close to a value of C6. Weight settings are listed in brackets following this format: [Odd, Pair, Quartr, Half].

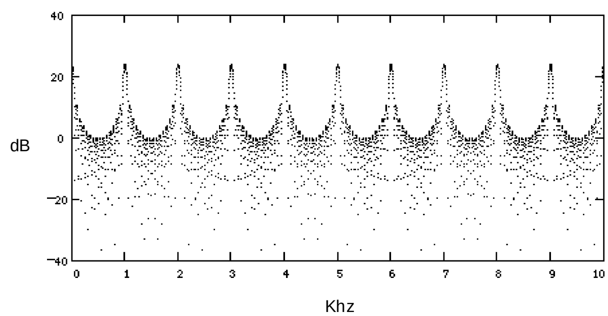


Figure 10-62 [100, 100, 100, 100]

In Figure 10-62, all peaks are exact multiples of the fundamental frequency set by the Pitch parameter. This setting gives the most "pitchiness" to the output.

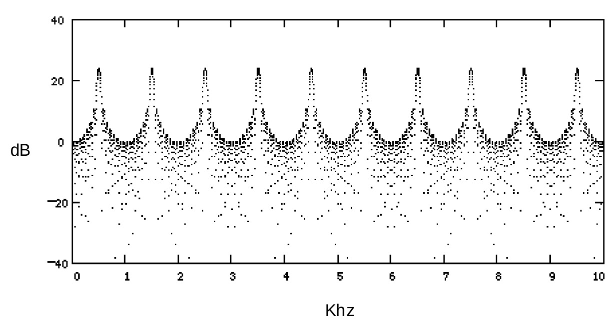


Figure 10-63 [-100, 100, 100, 100]

In Figure 10-63, peaks are odd multiples of a frequency one octave down from the Pitch setting. This gives a hollow, square-wavey sound to the output.

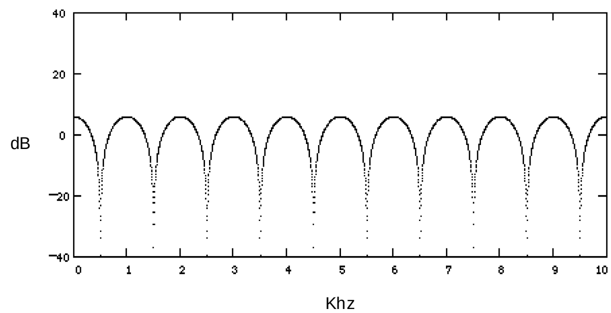


Figure 10-64 [100, 0, 0, 0]

In Figure 10-64, there are deeper notches between wider peaks

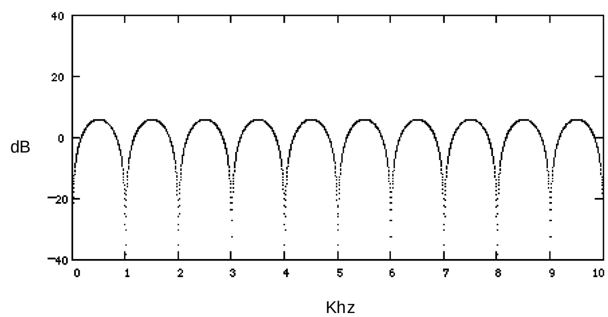


Figure 10-65 [-100, 0, 0, 0]

In Figure 10-65, there are peaks on odd harmonic multiples and notches on even harmonic multiples of a frequency one octave down from the Pitch setting.

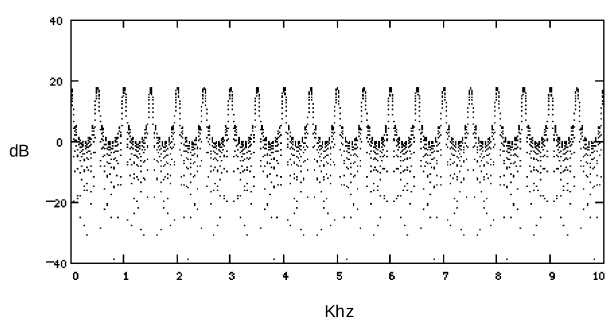


Figure 10-66 [0, 100, 100, 100]

Figure 10-66 is like [100,100,100,100], except that all the peaks are at (all) multiples of half the Pitch frequency.

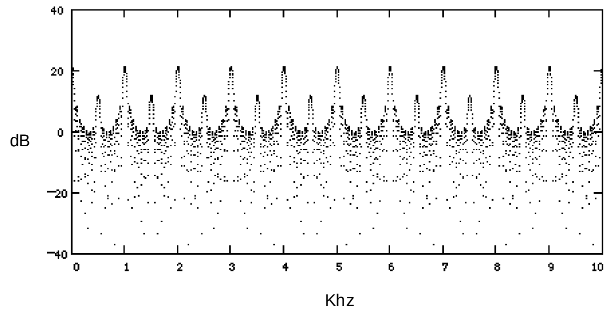


Figure 10-67 [50,100,100,100]

Figure 10-67 is halfway between [0,100,100,100] and [100,100,100,100].

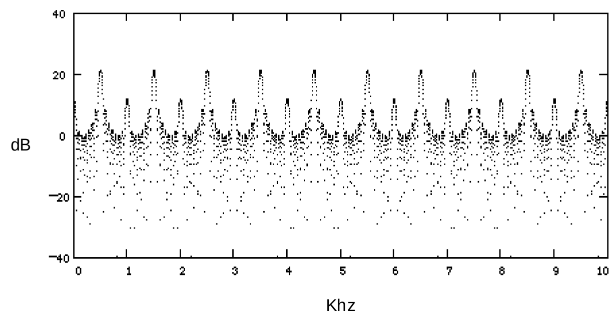


Figure 10-68 [-50,100,100,100]

Figure 10-68 is halfway between [0,100,100,100] and [-100,100,100,100]. If the Odd parameter is modulated with an FXMOD, then one can morph smoothly between the [100,100,100,100] and [-100,100,100,100] curves.

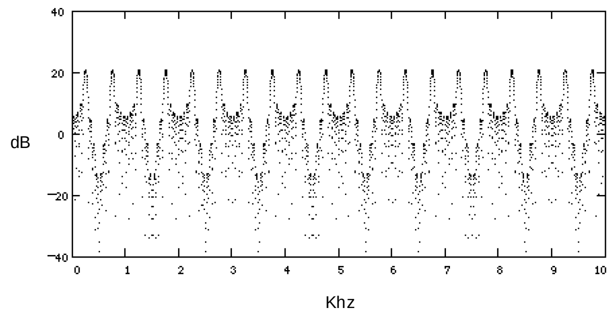


Figure 10-69 [100, -100, 100, 100]

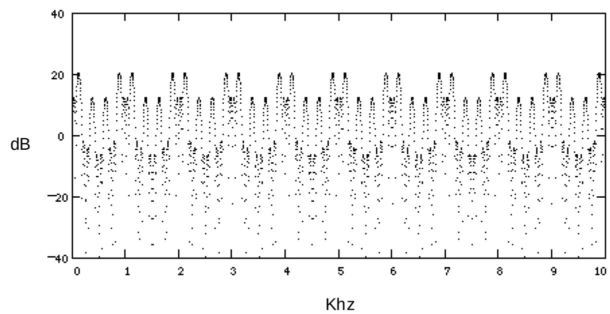


Figure 10-70 [100, 100, -100, 100]

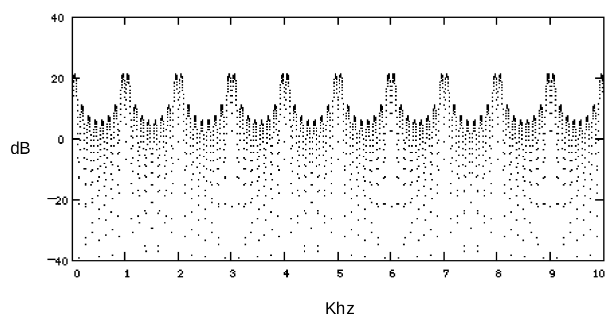


Figure 10-71 [100, 100, 100, -100]

The other 1,632,240,792 response curves have been omitted to save space.

Parameters

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Pitch	C-1 to G9	Ptch Offst	-12.0 to 12.0 ST
Odd Wts	-100 to 100 %	Quartr Wts	-100 to 100 %
Pair Wts	-100 to 100 %	Half Wts	-100 to 100 %

Wet/Dry	The relative amount of input signal and effected signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet.
Out Gain	The overall gain or amplitude at the output of the effect.
Pitch	The fundamental pitch imposed upon the input. Values are in MIDI note numbers.
Ptch Offst	An offset from the pitch frequency in semitones. This is also available for adding an additional continuous controller mod like pitch bend.
All other parameters	These parameters control the exact shape of the frequency response of Pitcher. An exact description of what each one does is, unfortunately, impossible, since there is a great deal of interaction between them. For examples, examine the figures above.

909 Super Shaper

Ridiculous shaper

PAUs: 1

The Super Shaper algorithm packs 2-1/2 times the number of shaping loops, and 8 times the gain of the VAST shaper. Refer to the section on shapers in the *Musician's Guide* for an overview of VAST shaper.

Setting Super Shaper amount under 1.00x produces the same nonlinear curve as that found in the VAST shaper. At values above 1.00x where the VAST shaper will pin at zero, the Super Shaper provides 6 more sine intervals before starting to zero-pin at 2.50x. The maximum shaper amount for Super Shaper is 32.00x.

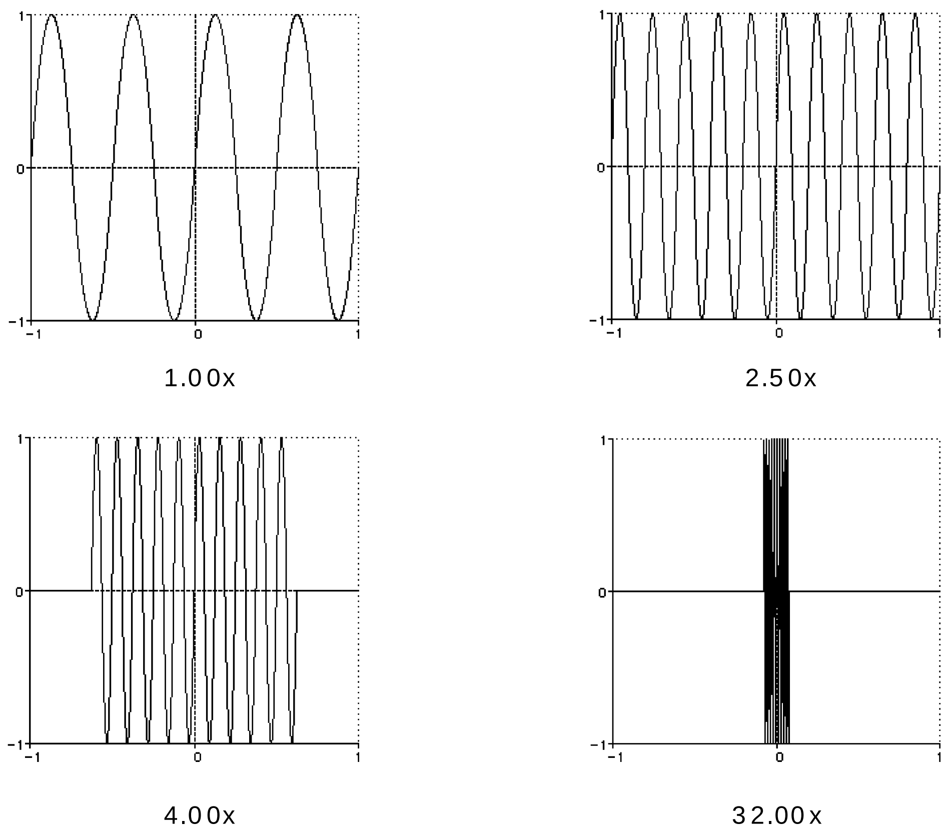


Figure 10-72 Super Shaper: Four Values of the Amount Parameter

Parameters

Wet/Dry	-100 to 100%	Out Gain	Off, -79.0 to 24.0 dB
Amount	0.10 to 32.00 x		

Wet/Dry The relative amount of input signal and effected signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet. Negative values polarity invert the wet signal.

Out Gain The overall gain or amplitude at the output of the effect.

Amount Adjusts the shaper intensity.

910 3 Band Shaper

3 band shaper

PAUs: 2

The 3 Band Shaper non-destructively splits the input signal into 3 separate bands using 1 pole (6dB/oct) filters, and applies a VAST-type shaper to each band separately. Refer to the Musicians Guide for an overview of VAST shaping. The cutoff frequencies for these filters are controlled with the CrossOver1 and CrossOver2 parameters. The low band contains frequencies from 0 Hz (dc) to the lower of the 2 CrossOver settings. The mid band contains frequencies between the 2 selected frequencies, and the hi band contains those from the higher of the 2 CrossOver settings all the way up to 24kHz.

Each frequency band has an enable switch for instantly bypassing any processing for that band , and a Mix control for adjusting the level of each band that is mixed at the output. negative Mix values polarity invert that band. The shaper Amt controls provide the same type of shaping as VAST shapers, but can go to 6.00x.

Parameters

Page 1

Wet/Dry	-100 to 100%	Out Gain	Off, -79.0 to 24.0 dB
CrossOver1	17 to 25088 Hz		
CrossOver2	17 to 25088 Hz		

Page 2

Lo Enable	On or Off	Lo Enable	On or Off
Lo Amt	0.10 to 6.00x	Lo Amt	0.10 to 6.00x
Lo Mix	-100 to 100%	Lo Mix	-100 to 100%
Mid Enable	On or Off		
Mid Amt	0.10 to 6.00x		
Mid Mix	-100 to 100%		

In/Out When set to “In” the effect is active; when set to “Out” the effect is bypassed.

Out Gain The overall gain or amplitude at the output of the effect.

CrossOver1 Adjusts one of the -6dB crossover points at which the input signal will be divided into the high, mid and low bands.

CrossOver2 Adjusts the other -6dB crossover points at which the input signal will be divided into the high, mid and low bands.

Enable Low, Mid, and High. Turns processing for each band on or off. Turning each of the 3 bands Off results in a dry output signal.

Amt Low, Mid, and High. Adjusts the shaper intensity for each band.

Mix Low, Mid, and High. Adjusts the level that each band is summed together as the wet signal. Negative values polarity invert the particular bands signal.

911 Mono LaserVerb
912 LaserVerb Lite
913 LaserVerb

A bizarre reverb with a falling buzz

PAUs: 1 for Mono LaserVerb
2 for LaserVerb Lite
3 for LaserVerb

LaserVerb is a new kind of reverb sound that has to be heard to be believed! When it is fed an impulsive sound such as a snare drum, LaserVerb plays the impulse back as a delayed train of closely spaced impulses, and as time passes, the spacing between the impulses gets wider. The close spacing of the impulses produces a discernable buzzy pitch which gets lower as the impulse spacing increases. The following figure is a simplified representation of the LaserVerb impulse response. (An impulse response of a system is what you would see if you had an oscilloscope on the system output and you gave the system an impulse or a spike for an input.)

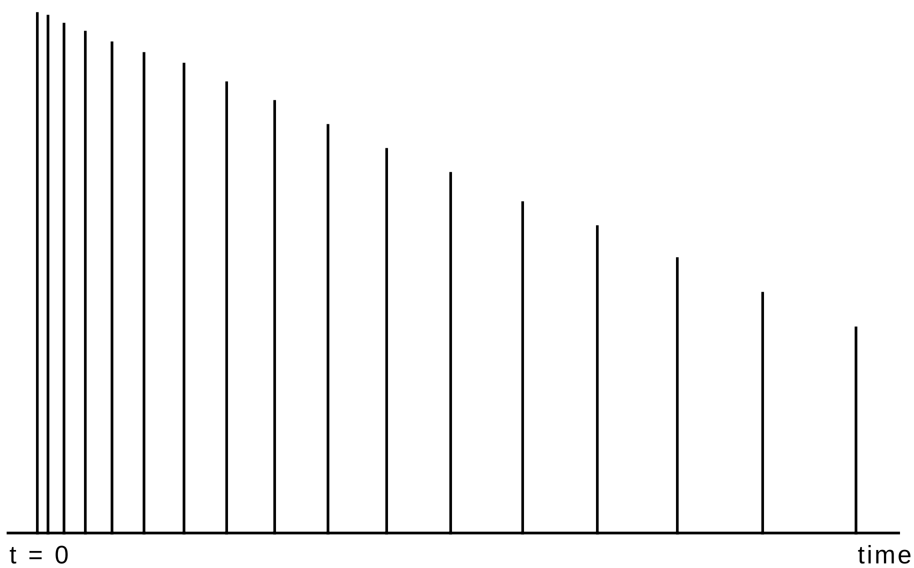


Figure 10-73 Simplified Impulse Response of LaserVerb

With appropriate parameter settings this effect produces a decending buzz or whine somewhat like a diving airplane or a siren being turned off. The descending buzz is most prominent when given an impulsive input such as a drum hit. When used as a reverb, it tends to be highly metallic and has high pitched tones at certain parameter settings. To get the decending buzz, start with about half a second of delay, set the Contour parameter to a high value (near 1), and set the HF Damping to a low value (at or near 0). The Contour parameter controls the overall shape of the LaserVerb impulse response. At high values the response builds up very quickly decays slowly. As the Coutour value is reduced, the decay becomes shorter and the sound takes longer to build up. At a setting of zero, the response degenerates to a simple delay.

The Spacing parameter controls the initial separation of impulses in the impulse response and the rate of their subsequent separation. Low values result in a high initial pitch (impulses are more closely spaced) and takes longer for the pitch to lower.

The output from LaserVerb can be fed back to the input. By turning up the feedback, the duration of the LaserVerb sound can be greatly extended. Cross-coupling may also be used to move the signal between left and right channels, producing a left/right ping-pong effect at the most extreme settings.

The 2 processing allocation unit (PAU) version is a sparser version than the 3 PAU version. It's buzzing is somewhat coarser. The 1 PAU version is like the 2 PAU version except the two input channels are summed and run through a single mono LaserVerb. The 1 PAU version does not have the cross-coupling control but does have output panning.

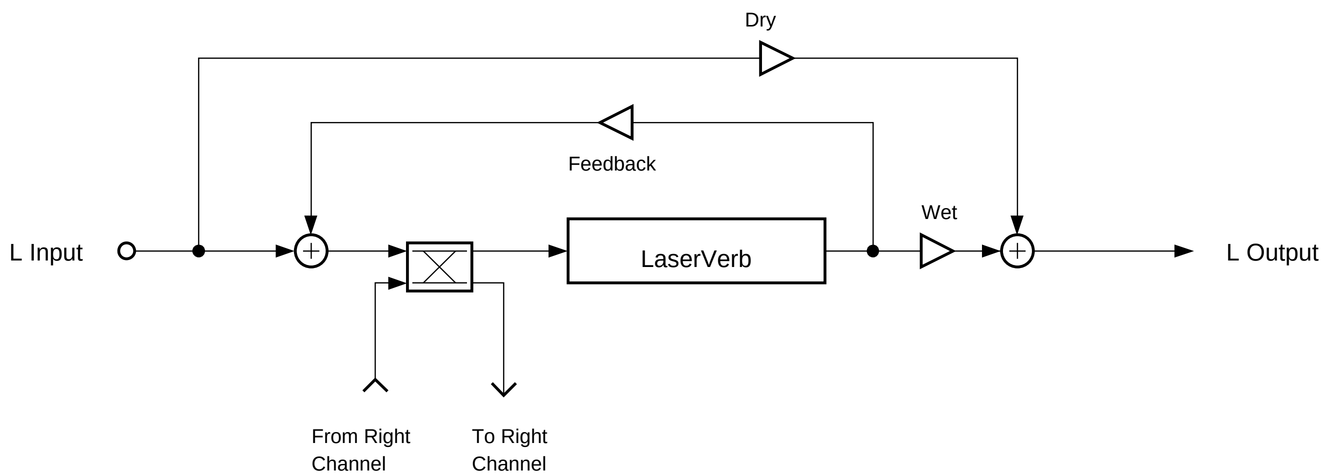


Figure 10-74 LaserVerb

Parameters for LaserVerb and LaserVerb Lite

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0dB
Fdbk Lvl	0 to 100%		
Xcouple	0 to 100%		
HF Damping	16 to 25088Hz		

Parameters for Mono LaserVerb

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0dB
Fdbk Lvl	0 to 100%	Pan	-100 to 100%
HF Damping	16 to 25088Hz		

Page 2

Dly Coarse	0 to 5000ms	Contour	0.0 to 100.0%
Dly Fine	-20.0 to 20.0ms		
Spacing	0.0 to 40.0samp		

Wet/Dry The amount of reverbed (wet) signal relative to unaffected (dry) signal.

Out Gain	The overall gain or amplitude at the output of the effect.
Fdbk Lvl	The percentage of the reverb output to feed back or return to the reverb input. Turning up the feedback is a way to stretch out the duration of the reverb, or, if the reverb is set to behave as a delay, to repeat the delay. The higher feedback is set, the longer the decay or echo will last.
Xcouple	LaserVerb & LaserVerb Lite are stereo effects. The cross-coupling control lets you send the sum of the input and feedback from one channel to its own LaserVerb effect (0% cross coupling) or to the other channel's effect (100% cross coupling) or somewhere in between. This control is not available in Mono LaserVerb.
HF Damping	The damping of high frequencies relative to low frequencies. When set to the highest frequency (25088 Hz), there is no damping and all frequencies decay at the same rate. At lower frequency settings, high frequency signal components will decay faster than low frequency components. If set too low, everything will decay almost immediately.
Pan	The Pan control is available in the Mono LaserVerb. The left and right inputs get summed to mono, the mono signal passes through the LaserVerb, and the final mono output is panned to the left and right outputs. Panning ranges from -100% (fully left), through 0% (centered), through to 100% (fully right).
Dly Coarse	You can set the overall delay length from 0 to 2 seconds (3 PAU) or 0 to 1.3 seconds (2 PAU). Lengthening the delay will increase the duration or decay time of the reverb. To reduce LaserVerb to a simple delay, set the Contour and Feedback controls to 0. Use a delay of about half a second as a starting point.
Dly Fine	The delay fine adjust is added to the delay coarse adjust to provide a delay resolution down to 0.1 ms.
Spacing	Determines the starting pitch of the decending buzz and how fast it decends. The Spacing parameter sets the initial separation of impulses in the impulse response and subsequent rate of increasing impulse separation. The spacing between impulses is given in samples and may be a fraction of a sample. (A sample is the time between successive digital words which is 20.8 μs or 1/48000 seconds.) For low values, the buzz starts at high frequencies and drops slowly. At high values the buzz starts at a lower pitch and drops rapidly.
Contour	Controls the overall envelope shape of the reverb. When set to a high value, sounds passed through the reverb start at a high level and slowly decay. As the control value is reduced, it takes some time for the effect to build up before decaying. At a value of around 34, the reverb is behaving like a reverse reverb, building up to a hit. When the Contour is set to zero, LaserVerb is reduced to a simple delay.

950 HardKnee Compress
951 SoftKneeCompress

Stereo hard- and soft-knee signal compression algorithms

PAUs: 1

The stereo hard- and soft-knee compressors are very similar algorithms and provide identical parameters and user interface. Both algorithms compress (reduce) the signal level when the signal exceeds a threshold. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the sum of the magnitudes of the left and right channels. A compression ratio of 1:1 will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and 1:1, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where 0 dBFS is digital full-scale and all other available values are negative.

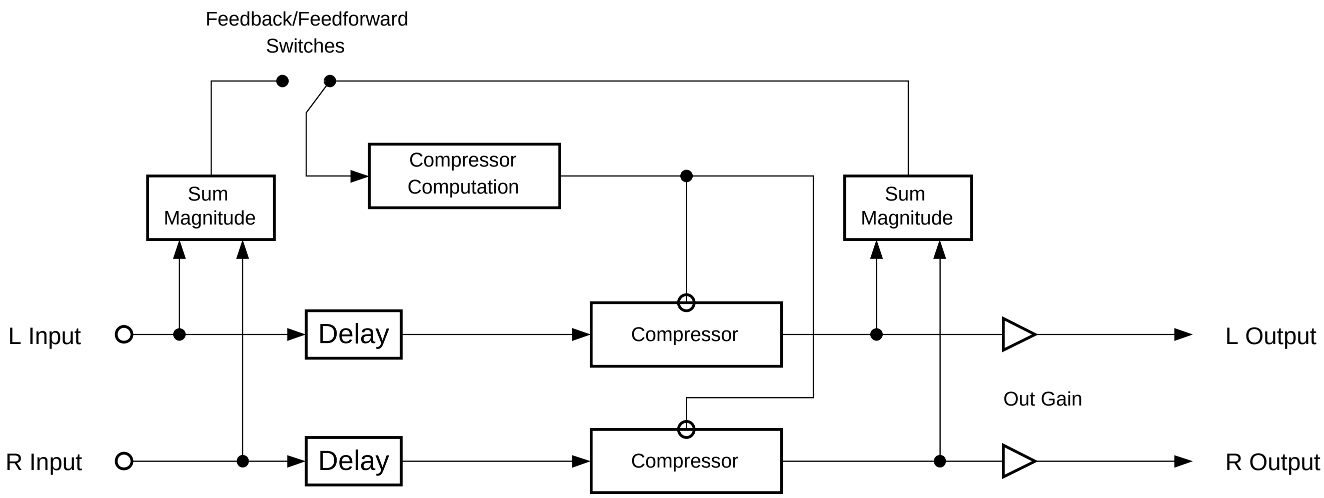


Figure 10-75 Compressor

In the hard-knee compressor, there is a sudden transition from uncompressed to compressed at the compression threshold. In the soft-knee compressor there is a more gradual transition from compressed to unity gain.

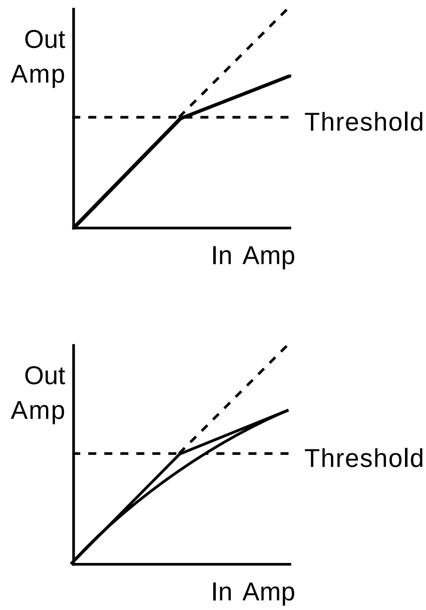


Figure 10-76 Hard- and Soft-Knee Compression Characteristics

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may over-shoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behaviour, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: “SmoothTime”. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to “Out”; for feedback compression, set it to “In”. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

In the feedback configuration, the signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing “knows” what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens. In the feed-forward configuration, the delay affects both the main signal and the side chain, and so is of limited usefulness. In compressors which use more than 1 PAU, the delay affects the main signal only, regardless of the side chain configuration.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of compression.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
FdbkComprs	In or Out		

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1.0:1 to 100:1, Inf:1
Rel Time	0 to 3000 ms	Threshold	-79.0 to 0.0dB
SmoothTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
Signal Dly	0.0 to 25.0ms		

In/Out When set to “In” the compressor is active; when set to “Out” the compressor is bypassed.

Out Gain Compressing the signal causes a reduction in signal level. To compensate, the output gain parameter may be used to increase the gain by as much as 24 dB. Note that the Out Gain parameter does not control the signal level when the algorithm is set to “Out”.

FdbkComprs A switch to set whether the compressor side chain is configured for feed-forward (Out) or feedback (In).

Atk Time The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Rel Time The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the expander’s envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly For the feed-forward setting, Signal Dly is the time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain pre-delay). This allows the compression to appear to take effect just before the signal actually rises. For feedback compression, this parameter causes both the side-chain and main signal path to be delayed together for limited benefit.

Ratio The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

Threshold The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

952 Expander

A stereo expansion algorithm

PAUs: 1

This is a stereo expander algorithm. The algorithms expands the signal (reduced the signal's gain) when the signal falls below the expansion threshold. The amount of expansion is based on the larger magnitude of the left and right channels. The amount of expansion is expressed as an expansion ratio. Expanding a signal reduces its level below the threshold. The expansion ratio is the inverse of the slope of the expander input/output characteristic. An expansion ratio of 1:1 will have no effect on the signal. A zero ratio (1:∞), will expand all signal levels below the threshold level to the null or zero level. (This expander expands to 1:17 at most.) Thresholds are expressed as a decibel level relative to digital full-scale (dBFS) where 0 dBFS is digital full-scale and all other available values are negative.

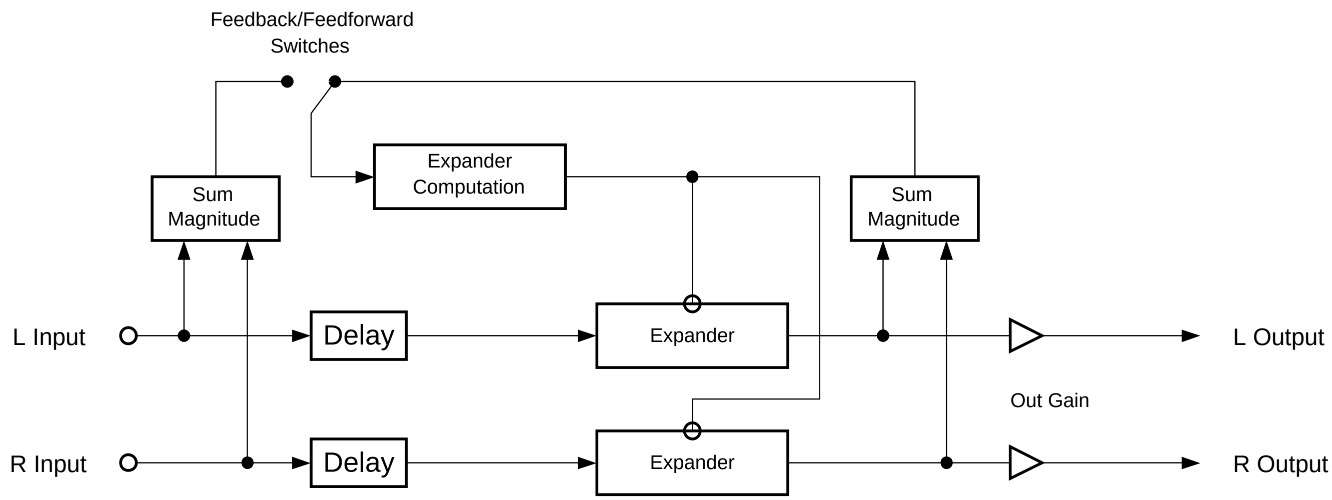


Figure 10-77 Expander

To determine how much to expand the signal, the expander must measure the signal level. Since musical signal levels will change over time, the expansion amounts must change as well. You can control how fast the expansion changes in response to changing signal levels with the attack and release time controls.

The attack time is defined as the time for the expansion to turn off when the signal rises above the threshold. This time should be very short for most applications. The expander release time is the time for the signal to expand down after the signal drops below threshold. The expander release time may be set quite long. An expander may be used to suppress background noise in the absence of signal, thus typical expander settings use a fast attack (to avoid losing real signal), slow release (to gradually fade out the noise), and the threshold set just above the noise level. You can set just how far to drop the noise with the expansion ratio.

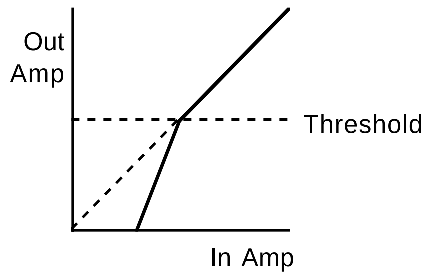


Figure 10-78 Expansion Transfer Characteristic

The signal being expanded may be delayed relative to the side chain processing. The delay allows the signal to stop being expanded just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by releasing the expander before the attack actually happens.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of expansion.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1:1.0 to 1:17.0
Rel Time	0 to 3000 ms	Threshold	-79.0 to 0.0 dB
SmoothTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
Signal Dly	0.0 to 25.0 ms		

In/Out When set to "In" the expander is active; when set to "Out" the expander is bypassed.

Out Gain The output gain parameter may be used to increase the gain by as much as 24 dB, or reduce the gain to nothing. Note that the Out Gain parameter does not control the signal level when the algorithm is set to "Out".

Atk Time The time for the expander to increase the gain of the signal (turns off the expander) after the signal rises above threshold.

Rel Time The time for the expander to reduce the signal level when the signal drops below the threshold (turning on expansion).

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the expander's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly The time in ms by which the input signal should be delayed with respect to expander side chain processing (i.e. side chain pre-delay). This allows the expansion to appear to turn off just before the signal actually rises.

Ratio The expansion ratio. High values (1:17 max) are highly expanded, low values (1:1 min) are moderately expanded.

Threshold The expansion threshold level in dBFS (decibels relative to full scale) below which the signal begins to be expanded.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to expansion.

953 Compress w/SC EQ

Stereo soft-knee compression algorithm with filtering in the side chain

PAUs: 2

The Compress w/SC EQ algorithm is the same as the SoftKneeCompress algorithm except that equalization has been added to the side chain signal path. The equaliztion to the side chain includes bass and treble shelf filters and a parametric mid-range filter.

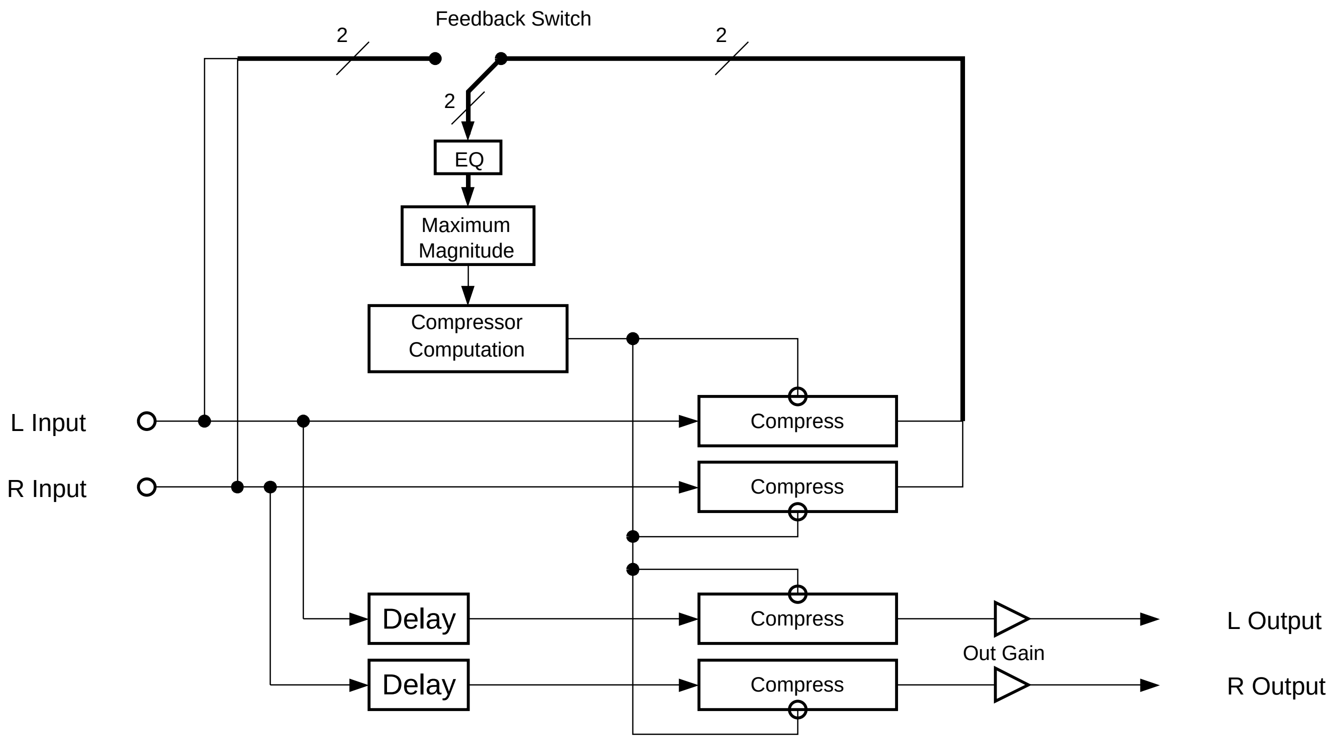


Figure 10-79 Compressor with side chain equalization.

Using side chain equalization allows you to compress your signal based on the spectral (frequency) content of your signal. For example, by boosting the treble shelf filter, you can compress the signal only when there is a lot of high frequencies present.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
FdbkComprs	In or Out		

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1.0:1 to 100.0:1, Inf:1
Rel Time	0 to 3000 ms	Threshold	-79.0 to 24.0 dB
SmoothTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
Signal Dly	0.0 to 25.0 ms		

Page 3

SCBassGain	-79.0 to 24.0 dB	SCTrebGain	-79.0 to 24.0 dB
SCBassFreq	16 to 25088 Hz	SCTrebFreq	16 to 25088 Hz
SCMidGain	-79.0 to 24.0 dB		
SCMidFreq	16 to 25088 Hz		
SCMidWidth	0.010 to 5.000 oct		

In/Out When set to "In" the compressor is active; when set to "Out" the compressor is bypassed.

Out Gain Compressing the signal causes a reduction in signal level. To compensate, the output gain parameter may be used to increase the gain by as much as 24 dB. Note that the Out Gain parameter does not control the signal level when the algorithm is set to "Out".

FdbkComprs A switch to set whether the compressor side chain is configured for feed-forward (Out) or feedback (In).

Atk Time The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Rel Time The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the expander's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain pre-delay). This allows the compression to appear to take effect just before the signal actually rises.

Ratio The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

Threshold The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

SCBassGain The amount of boost or cut that the side chain bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

SCBassFreq The center frequency of the side chain bass shelving filter in intervals of one semitone.

SCTrebGain The amount of boost or cut that the side chain treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

SCTrebFreq The center frequency of the side chain treble shelving filters in intervals of one semitone.

SCMidGain The amount of boost or cut that the side chain parametric mid filter should apply in dB to the specified frequency band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

SCMidFreq The center frequency of the side chain parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

SCMidWidth The bandwidth of the side chain parametric mid filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

954 Compress/Expand
955 Comp/Exp + EQ

A stereo soft-knee compression and expansion algorithm with and without equalization

PAUs: 2 for Compress/Expand
3 for Cmp/Exp + EQ

These are a stereo compressor and expander algorithms. One version is followed by equalization and the other is not. The algorithms compress the signal level when the signal exceeds a compression threshold and expands the signal when the signal falls below the expansion threshold. The amount of compression and/or expansion is based on the larger magnitude of the left and right channels.

Compression is expressed as a ratio: the inverse of the slope of the compressor input/output characteristic. A compression ratio of 1:1 has no effect on the signal. An infinite ratio compresses all signal levels above the threshold level to the threshold level (zero slope). For ratios between infinite and 1:1, increasing the input will increase the output, but by less than it would without compression. The compressor is a soft-knee compressor, so the transition from compressed to linear is gradual.

The amount of expansion is expressed as an expansion ratio. Expanding a signal reduces its level below the threshold. The expansion ratio is the inverse of the slope of the expander input/output characteristic. An expansion ratio of 1:1 will have no effect on the signal. A zero ratio (1:∞), will expand all signal levels below the threshold level to the null or zero level. (This expander expands to 1:17 at most.) Thresholds are expressed as a decibel level relative to digital full-scale (dBFS) where 0 dBFS is digital full-scale and all other available values are negative.

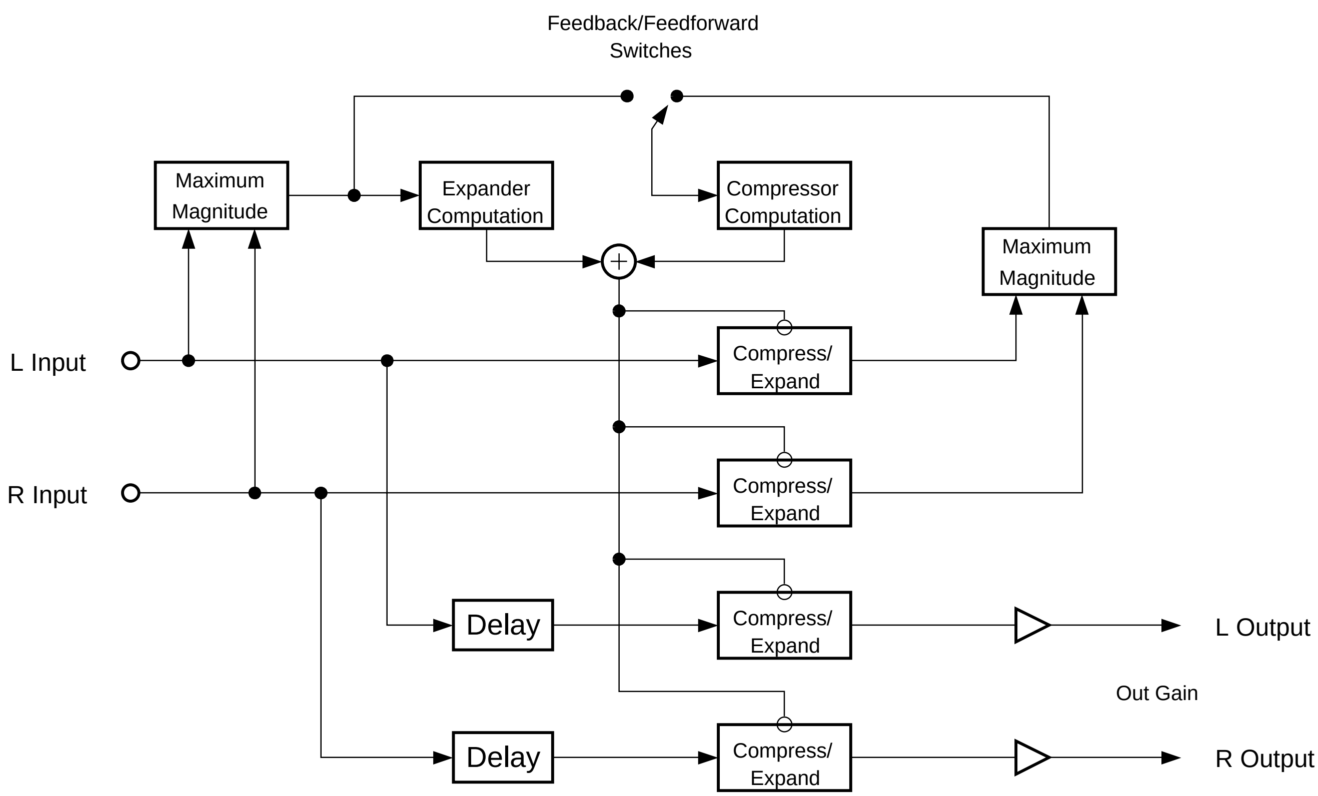


Figure 10-80 Compressor/Expander (optional EQ not shown)

To determine how much to compress or expand the signal, the compressor/expander must measure the signal level. Since musical signal levels will change over time, the compression and expansion amounts must change as well. You can control how fast the compression or expansion changes in response to changing signal levels with the attack and release time controls. Compression and expansion have separate controls.

First consider the compressor. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may over-shoot the threshold level for some time interval before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behaviour, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: “SmoothTime”. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release times. Generally the smoothing time should be kept at or shorter than the attack time.

This compressor provides two compressed segments. The signal below the lower threshold is not compressed. The compression ratio corresponding to the lower threshold sets the amount of compression for the lower compression segment. Above the upper threshold, the signal is compressed even further by the ratio corresponding to the upper threshold. You may use the upper segment as a limiter (infinite compression), or you may use the two compression segments to produce compression with a softer knee than you would get otherwise. For example, to make the algorithm a compressor and limiter, first choose the two thresholds. The limiter will of course have the higher threshold. Set the compression ratio for the higher threshold to “Inf:1”. This gives you your limiter. Finally set the compression ratio for the lower threshold to the amount of compression that you want. Either pair of threshold and ratio parameters may be used for the upper compression segment -- they are interchangeable. Above the upper threshold, the two compression ratios become additive. If both ratios are set to 3.0:1, then the compression of the upper segment will be 6.0:1. Another way to think of it is as two compressors wired in series (one after the other).

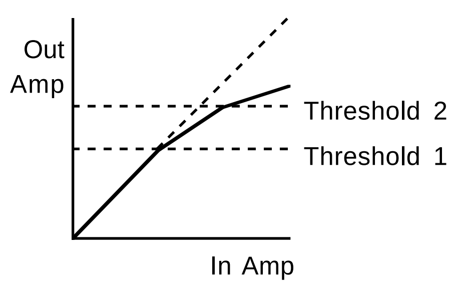


Figure 10-81 Two Segment Compression Characteristic

You have the choice of using the compressor configured as feed-forward or feedback. For feed-forward, set the FdbkComprs parameter to “Out”; for feedback compression, set it to “In”. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

The expander attack/release times are similar, though there is only one expand segment. The expander works independently of the compressor. The expander cannot be configured for feedback (if it could, it would alway shut itself off permanently). The signal delay path does affect the expander. The attack time is defined as the time for the expansion to turn off when the signal rises above the threshold. This time should be very short for most applications. The expander release time is the time for the signal to expands down after the signal drops below threshold. The expander release time may be set quite long. An expander may be used to suppress background noise in the absence of signal, thus typical expander settings use a fast attack (to avoid losing real signal), slow release (to gradually fade out the noise), and the threshold set just above the noise level. You can set just how far to drop the noise with the expansion ratio.

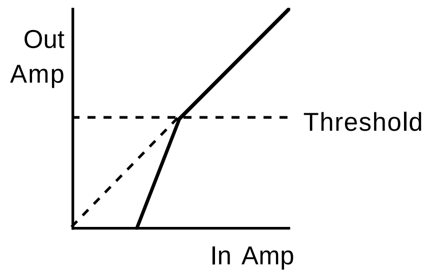


Figure 10-82 Expansion Transfer Characteristic

The signal being compressed/expanded may be delayed relative to the side chain processing. The delay allows the signal to start being compressed (or stop being expanded) just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens (or releasing the expander before the attack happens). This feature works whether the side chain is configured for feed-forward or feedback.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of compression and expansion.

The algorithm Comp/Exp + EQ differs from Compress/Expand in that the compressor and expander sections are followed by equalization filters. The output signal may be filtered with bass and treble shelving filters and a mid-range parameteric filter.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
FdbkComprs	In or Out		

Page 2

Comp Atk	0.0 to 228.0 ms	Exp Atk	0.0 to 228.0 ms
Comp Rel	0 to 3000 ms	Exp Rel	0 to 3000 ms
SmoothTime	0.0 to 228.0 ms		
Signal Dly	0.0 to 25.0 ms		

Page 3

Comp1Ratio	1.0:1 to 100.0:1, Inf:1	Exp Ratio	1:1.0 to 1:17.0
Comp1Thres	-79.0 to 0.0 dB	Exp Thres	-79.0 to 0.0 dB
Comp2Ratio	1.0:1 to 100.0:1, Inf:1	MakeUpGain	Off, -79.0 to 24.0 dB
Comp2Thres	-79.0 to 0.0 dB		

Page 4

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	16 to 25088 Hz	Treb Freq	16 to 25088 Hz
Mid Gain	-79.0 to 24.0 dB		
Mid Freq	16 to 25088 Hz		
Mid Wid	0.010 to 5.000 oct		

In/Out When set to “In” the compressor/expander is active; when set to “Out” the compressor/expander is bypassed.

Out Gain Compressing the signal causes a reduction in signal level. To compensate, the output gain parameter may be used to increase the gain by as much as 24 dB. Note that the Out Gain parameter does not control the signal level when the algorithm is set to “Out”.

FdbkComprs A switch to set whether the compressor side chain is configured for feed-forward (Out) or feedback (In). The expander is unaffected.

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

Exp Atk The time for the expander to increase the gain of the signal (turns off the expander) after the signal rises above threshold.

Exp Rel The time for the expander to reduce the signal level when the signal drops below the threshold (turning on expansion).

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the expander’s envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain pre-delay). This allows the compression to appear to take effect just before the signal actually rises.

Comp1Ratio The compression ratio in effect above compression threshold #1 (Comp1Thres). High ratios are highly compressed; low ratios are moderately compressed.

Comp1Thres One of two compression threshold levels. Threshold is expressed in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

Comp2Ratio The compression ratio in effect above compression threshold #2 (Comp2Thres). High ratios are highly compressed; low ratios are moderately compressed.

Comp2Thres One of two compression threshold levels. Threshold is expressed in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

Exp Ratio The expansion ratio. High values (1:17 max) are highly expanded, low values (1:1 min) are moderately expanded.

Exp Thres The expansion threshold level in dBFS (decibels relative to full scale) below which the signal begins to be expanded.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression or expansion.

Bass Gain The amount of boost or cut that the bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency. [Comp/Exp + EQ only]

Bass Freq The center frequency of the bass shelving filter in intervals of one semitone. [Comp/Exp + EQ only]

Treb Gain The amount of boost or cut that the treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency. [Comp/Exp + EQ only]

Treb Freq The center frequency of the treble shelving filter in intervals of one semitone. [Comp/Exp + EQ only]

Mid Gain The amount of boost or cut that the mid parametric filter should apply in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency. [Comp/Exp + EQ only]

Mid Freq The center frequency of the mid parametric filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency. [Comp/Exp + EQ only]

Mid Wid The bandwidth of the mid parametric filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response. [Comp/Exp + EQ only]

956 Compress 3 Band

Stereo soft-knee 3 frequency band compression algorithm

PAUs: 4

The 3 band compressor divides the input stereo signal into 3 frequency bands and runs each band through its own stereo soft-knee compressor. After compression, the bands are summed back together to produce the output. You may set the frequencies at which the bands are split.

The compressors reduce the signal level when the signal exceeds a threshold. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the sum of the magnitudes of the left and right channels. A compression ratio of 1:1 will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and 1:1, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where 0 dBFS is digital full-scale and all other available values are negative.

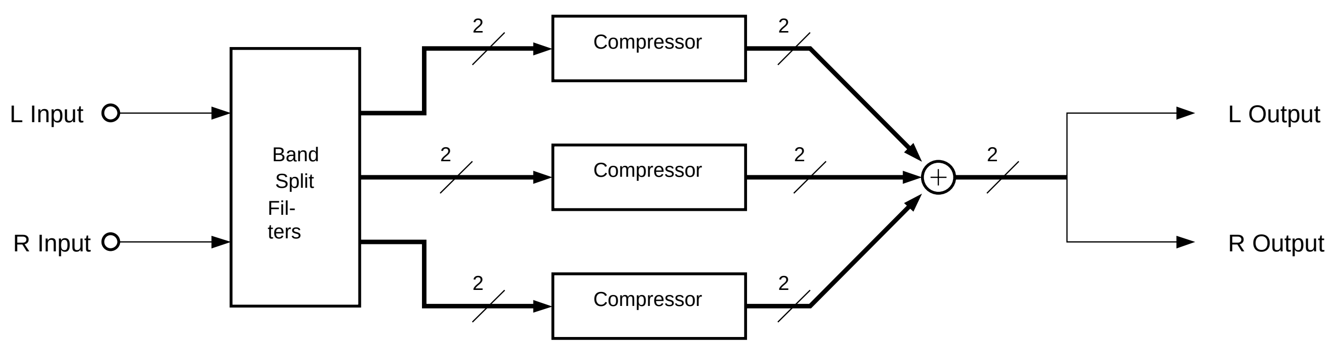


Figure 10-83 Band Compressor

In the soft-knee compressor there is a gradual transition from compressed to unity gain.

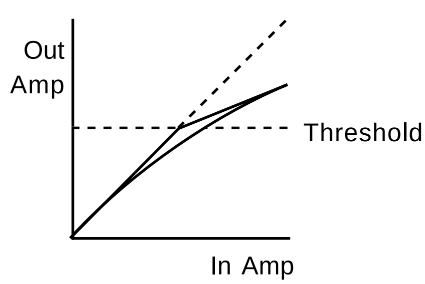


Figure 10-84 Soft-Knee Compression Characteristics

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may over-shoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behaviour, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: "Smth *Band*". The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to "Out"; for feedback compression, set it to "In". The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

The signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens. This feature works whether the side chain is configured for feed-forward or feedback.

A meter is provided for each compression band to display the amount of gain reduction that is applied to the signal as a result of compression.

*P*Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
FdbkComprs	In or Out	Crossover1	16 to 25088 Hz
Signal Dly	0.0 to 25.0 ms	Crossover2	16 to 25088 Hz

Page 2

Atk Low	0.0 to 228.0 ms	Ratio Low	1.0:1 to 100.0:1, Inf:1
Rel Low	0 to 3000 ms	Thres Low	-79.0 to 24.0 dB
Smth Low	0.0 to 228.0 ms	MakeUp Low	Off, -79.0 to 24.0 dB

Page 3

Atk Mid	0.0 to 228.0 ms	Ratio Mid	1.0:1 to 100.0:1, Inf:1
Rel Mid	0 to 3000 ms	Thres Mid	-79.0 to 24.0 dB
Smth Mid	0.0 to 228.0 ms	MakeUp Mid	Off, -79.0 to 24.0 dB

Page 4

Atk High	0.0 to 228.0 ms	Ratio High	1.0:1 to 100.0:1, Inf:1
Rel High	0 to 3000 ms	Thres High	-79.0 to 24.0 dB
Smth High	0.0 to 228.0 ms	MakeUpHigh	Off, -79.0 to 24.0 dB

In/Out When set to “In” the compressor is active; when set to “Out” the compressor is bypassed.

Out Gain Compressing the signal causes a reduction in signal level. To compensate, the output gain parameter may be used to increase the gain by as much as 24 dB. Note that the Out Gain parameter does not control the signal level when the algorithm is set to “Out”.

FdbkComprs A switch to set whether the compressor side chain is configured for feed-forward (Out) or feedback (In).

Signal Dly The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain pre-delay). This allows the compression to appear to take effect just before the signal actually rises.

Crossover*N* The Crossover parameters (1 and 2) set the frequencies which divide the three compression frequency bands. The two parameters are interchangeable, so either may contain the higher frequency value.

Atk Low, Mid or High. The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Rel Low, Mid, and High. The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

Smth Low, Mid, and High. A lowpass filter in the control signal path. It is intended to smooth the output of the expander’s envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Ratio Low, Mid, and High. The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

Thres Low, Mid, and High. The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

957 Gate
958 Super Gate

Signal gate algorithms

PAUs: 1 for Gate
2 for Super Gate

Gate and Super Gate do stand alone gate processing and can be configured as a stereo or mono effects. As a stereo effect, the stereo signal gates itself based on its amplitude. As a mono effect, you can use one mono input signal to gate a second mono input signal (or one channel can gate itself). Separate output gain and panning for both channels is provided for improved mono processing flexibility.

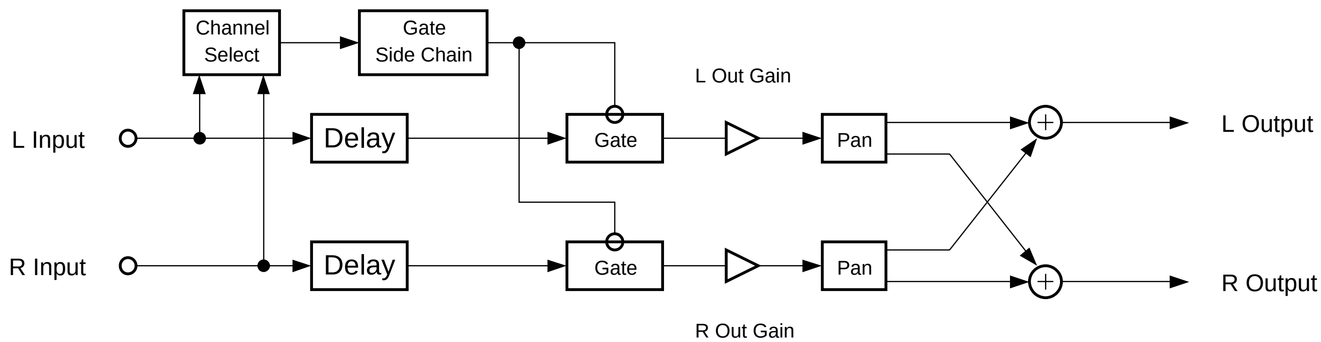


Figure 10-85 Gate

A gate behaves like an on off switch for a signal. One or both input channels is used to control whether the switch is on (gate is open) or off (gate is closed). The on/off control is called "side chain" processing. You select which of the two input channels or both is used for side chain processing. When you select both channels, the sum of the left and right input amplitudes is used. The gate is opened when the side chain amplitude rises above a level that you specify with the Threshold parameter.

Super Gate will behave differently depending on whether the Retrigger parameter is set to off or on. For the simpler Gate, there is no Retrigger parameter, and it is as if Retrigger is always on. If Retrigger is on, the gate will stay open for as long as the side chain signal is above the threshold. When the signal drops below the threshold, the gate will remain open for the time set with the Gate Time parameter. At the end of the Gate Time, the gate closes. When the signal rises above threshold, it opens again. What is happening is that the gate timer is being constantly retriggered while the signal is above threshold. You will typically use the gate with Retrigger set to on for percussive sounds.

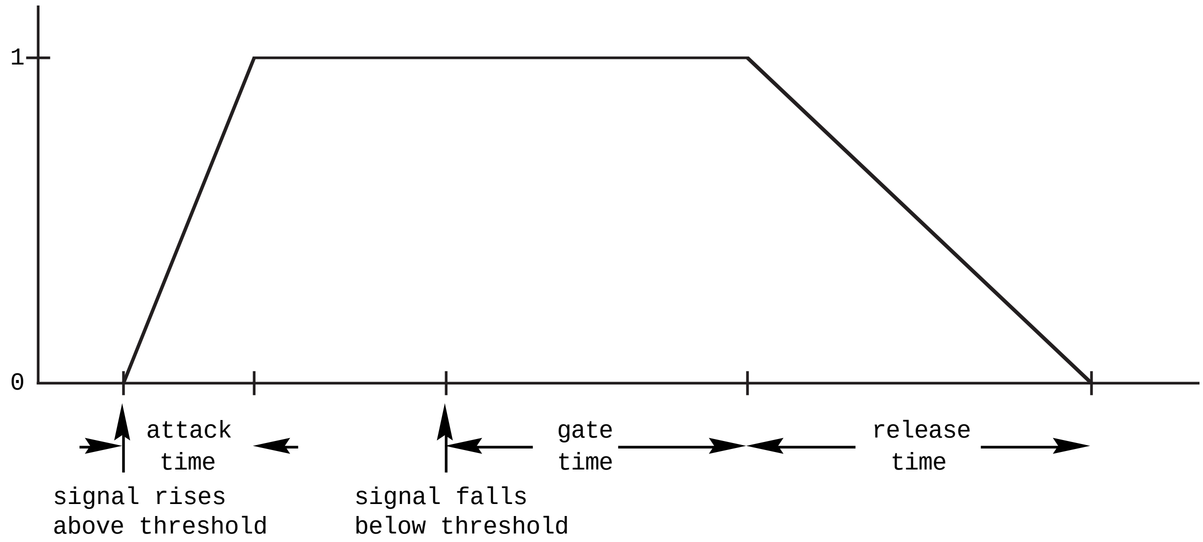


Figure 10-86 Signal envelope for Gate and Super Gate when Retrigger is "On"

If Retrigger is off (Super Gate only), then the gate will open when the side chain signal rises above threshold as before. The gate will then close as soon as the gate time has elapsed, whether or not the signal is still above threshold. The gate will not open again until the envelope of the side chain signal falls below the threshold and rises above threshold again. Since an envelope follower is used, you can control how fast the envelope follows the signal with the Env Time parameter. Retrigger set to off is useful for gating sustained sounds or where you need precise control of how long the gate should remain open.

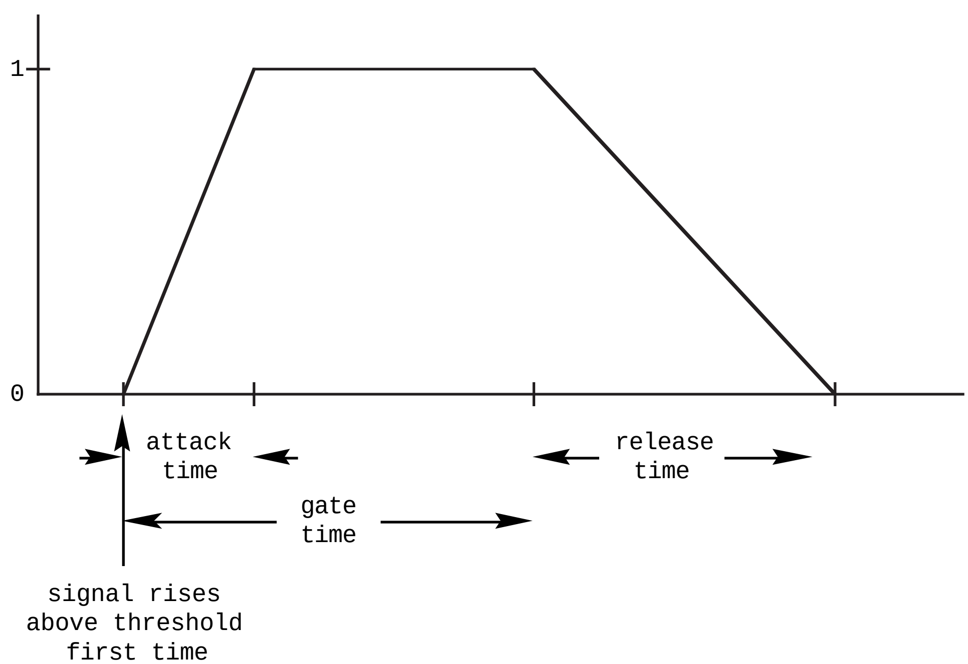


Figure 10-87 Super Gate signal envelope when Retrigger is "Off"

If Ducking is turned on, then the behaviour of the gate is reversed. The gate is open while the side chain signal is below threshold, and it closes when the signal rises above thresold.

If the gate opened and closed instantaneously, you would hear a large digital click, like a big knife switch was being thrown. Obviously that's not a good idea, so Atk Time (attack) and Rel Time (release) parameters are use to set the times for the gate to open and close. More precisely, depending on whether Ducking is off or on, Atk Time sets how fast the gate opens or closes when the side chain signal rises above the threshold. The Rel Time sets how fast the gate closes or opens after the gate timer has elapsed.

The Signal Dly parameter delays the signal being gated, but does not delay the side chain signal. By delaying the main signal relative to the side chain signal, you can open the gate just before the main signal rises above threshold. It's a little like being able to pick up the telephone before it rings!

For Super Gate (not the simpler Gate), filtering can be done on the side chain signal. There are controls for a bass shelf filter, a treble shelf filter and a parametric (mid) filter. By filtering the side chain, you can control the sensitivity of the gate to different frequencies. For example, you can have the gate open only if high frequencies are present -- or only if low frequencies are present.

Parameters for Gate

Page 1

In/Out	In or Out		
L Out Gain	Off, -79.0 to 24.0 dB	R Out Gain	Off, -79.0 to 24.0 dB
L Pan	-100 to 100%	R Pan	-100 to 100%
SC Input	(L+R)/2		

Page 2

Threshold	-79.0 to 24.0 dB	Gate Time	0 to 3000 ms
Ducking	On or Off	Atk Time	0.0 to 228.0 ms
Retrigger [Super]	On or Off	Rel Time	0 to 3000 ms
Env Time [Super]	0 to 3000 ms	Signal Dly	0.0 to 25.0 ms

Additional Parameters for Super Gate

Page 1

SCBassGain	-79.0 to 24.0 dB	SCTrebGain	-79.0 to 24.0 dB
SCBassFreq	16 to 25088 Hz	SCTrebFreq	16 to 25088 Hz
SCMidGain	-79.0 to 24.0 dB		
SCMidFreq	16 to 25088 Hz		
SCMidWidth	0.010 to 5.000 oct		

In/Out When set to "In" the gate is active; when set to "Out" the gate is bypassed.

L/R Out Gain The separate output signal levels in dB for the left and right channels. The output gains are calculated before the final output panning.

L/R Pan Both of the gated signal channels can be panned between left and right prior to final output. This can be useful when the gate is used as a mono effect, and you don't want to hear one of the input channels, but you want your mono output panned to stereo. -100% is panned to the left, and 100% is panned to the right.

SC Input The side chain input may be the amplitude of the left L input channel, the right R input channel, or the sum of the amplitudes of left and right (L+R)/2. You can gate a stereo signal with itself by using the sum, a mono signal with itself, or you can gate a mono signal using a second mono signal as the side chain.

Threshold The signal level in dB required to open the gate (or close the gate if Ducking is on).

Ducking When set to “Off”, the gate opens when the signal rises above threshold and closes when the gate time expires. When set to “On”, the gate closes when the signal rises above threshold and opens when the gate time expires.

Retrigger If Retrigger is “On”, the gate timer is constantly restarted (retriggered) as long as the side chain signal is above the threshold. The gate then remains open (assuming Ducking is “Off”) until the signal falls below the threshold and the gate timer has elapsed. If Retrigger is “Off”, then the gate timer starts at the moment the signal rises above the threshold and the gate closes after the timer elapses, whether or not the signal is still above threshold. With Retrigger off, use the Env Time to control how fast the side chain signal envelope drops below the threshold. With Retrigger set to off, the side chain envelope must fall below threshold before the gate can open again. [Super Gate only]

Env Time Envelope time is for use when Retrigger is set to “Off”. The envelope time controls the time for the side chain signal envelope to drop below the threshold. At short times, the gate can reopen rapidly after it has closed, and you may find the gate opening unexpectedly due to an amplitude modulation of the side chain signal. For long times, the gate will remain closed until the envelope has a chance to fall, and you may miss gating events.

Gate Time The time in seconds that the gate will stay fully on after the signal envelope rises above threshold. The gate timer is started or restarted whenever the signal envelope rises above threshold. If Retrigger is On, the gate timer is continually reset while the side chain signal is above the threshold.

Atk Time The time for the gate to ramp from closed to open (reverse if Ducking is on) after the signal rises above threshold.

Rel Time The time for the gate to ramp from open to closed (reverse if Ducking is on) after the gate timer has elapsed.

Signal Dly The delay in milliseconds (ms) of the signal to be gated relative to the side chain signal. By delaying the main signal, the gate can be opened before the main signal rises above the gating threshold.

Super Gate Parameters

SCBassGain The amount of boost or cut that the side chain bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

SCBassFreq The center frequency of the side chain bass shelving filters in intervals of one semitone.

SCTrebGain The amount of boost or cut that the side chain treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

SCTrebFreq The center frequency of the side chain treble shelving filters in intervals of one semitone.

SCMidGain The amount of boost or cut that the side chain parametric mid filter should apply in dB to the specified frequency band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

SCMidFreq The center frequency of the side chain parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

SCMidWidth The bandwidth of the side chain parametric mid filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

959 2 Band Enhancer

2 band spectral modifier

PAUs: 1

The 2 Band Enhancer modifies the spectral content of the input signal primarily by brightening signals with little or no high frequency content, and boosting pre-existing bass energy. First, the input is non-destructively split into 2 frequency bands using 6 dB/oct hipass and lopass filters (Figure 1). The hipassed band is processed to add additional high frequency content by using a nonlinear transfer function in combination with a high shelving filter. Each band can then be separately delayed to sample accuracy and mixed back together in varying amounts. One sample of delay is approximately equivalent to 20 microseconds, or 180 degrees of phase shift at 24 khz. Using what we know about psychoacoustics, phase shifting, or delaying certain frequency bands relative to others can have useful affects without adding any gain. In this algorithm, delaying the lopassed signal relative to the hipass signal brings out the high frequency transient of the input signal giving it more definition. Conversely, delaying the hipass signal relative to the lopass signal brings out the low frequency transient information which can provide punch.

The transfer applied to the hipass signal can be used to generate additional high frequency content when set to a non-zero value. As the value is scrolled away from 0, harmonic content is added in increasing amounts to brighten the signal. In addition to adding harmonics, positive values impose a dynamically compressed quality, while negative values sound dynamically expanded. This type of compression can bring out frequencies in a particular band even more. The expanding quality is particularly useful when trying to restore transient information.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
CrossOver	17 to 25088 Hz		

Page 2

Hi Drive	Off, -79.0 to 24.0 dB		
Hi Xfer	-100 to 100%		
Hi Shelf F	16 to 25088 Hz		
Hi Shelf G	-96 to 24 dB		
Hi Delay	0 to 500 samp	Lo Delay	0 to 500 samp
Hi Mix	Off, -79.0 to 24.0 dB	Lo Mix	Off, -79.0 to 24.0 dB

In/Out When set to “In” the effect is active; when set to “Out” the effect is bypassed.

Out Gain The overall gain or amplitude at the output of the effect.

CrossOver Adjusts the -6dB crossover point at which the input signal will be divided into the hipass band and a lopass bands.

Hi Drive Adjusts the gain into the transfer function. The affect of the transfer can be intensified or reduced by respectively increasing or decreasing this value.

Hi Xfer The intensity of the transfer function.

Hi Shelf F The frequency of where the high shelving filter starts to boost or attenuate.

Hi Shelf G	The boost or cut of the high shelving filter.
Hi Delay	Adjusts the number of samples the hipass signal is delayed.
Hi Mix	Adjusts the output gain of the hipass signal.
Lo Delay	Adjusts the number of samples the lopass signal is delayed.
Lo Mix	Adjusts the output gain of the lopass signal.

960 3 Band Enhancer

3 band spectral modifier

PAUs: 2

The 3 Band Enhancer modifies the spectral content of the input signal by boosting existing spectral content, or stimulating new ones. First, the input is non-destructively split into 3 frequency bands using 6 dB/oct hipass and lopass filters (Figure 1). The high and mid bands are separately processed to add additional high frequency content by using two nonlinear transfer functions. The low band is processed by a single nonlinear transfer to enhance low frequency energy. Each band can also be separately delayed to sample accuracy and mixed back together in varying amounts. One sample of delay is approximately equivalent to 20 microseconds, or 180 degrees of phase shift with the KDFX 24 khz sampling rate. Using what we know about psychoacoustics, phase shifting, or delaying certain frequency bands relative to others can have useful affects without adding any gain. In this algorithm, delaying the lower bands relative to higher bands brings out the high frequency transient of the input signal giving it more definition. Conversely, delaying the higher bands relative to the lower bands brings out the low frequency transient information which can provide punch.

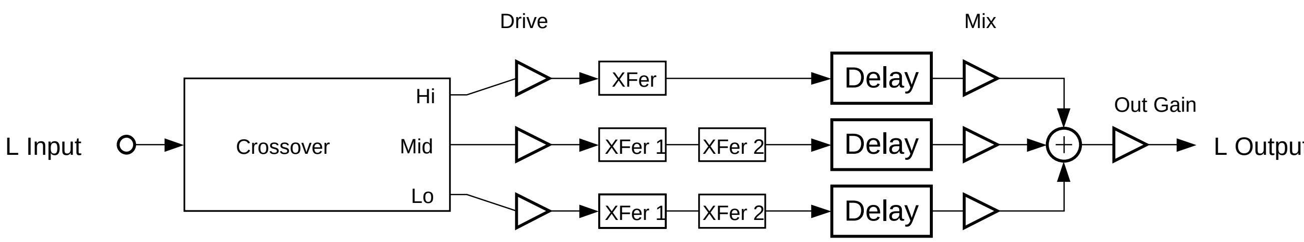


Figure 10-88 One channel of 3 Band Enhancer

The nonlinear transfers applied to the high and mid bands can be used to generate additional high and mid frequency content when Xfer1 and Xfer2 are set to non-zero values. As the value is scrolled away from 0, harmonic content is added in increasing amounts. In addition, setting both positive or negative will respectively impose a dynamically compressed or expanded quality. This type of compression can bring out frequencies in a particular band even more. The expanding quality is useful when trying to restore transient information. More complex dynamic control can be obtained by setting these independent of each other. Setting one positive and the other negative can even reduce the noise floor in some applications.

The low band has a nonlinear transfer that requires only one parameter. Its affect is controlled similarly.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
CrossOver1	17 to 25088 Hz		
CrossOver2	17 to 25088 Hz		

Page 2

Lo Enable	On or Off	Mid Enable	On or Off
Lo Drive	Off, -79.0 to 24.0 dB	Mid Drive	Off, -79.0 to 24.0 dB
Lo Xfer	-100 to 100%	Mid Xfer1	-100 to 100%
		Mid Xfer2	-100 to 100%
Lo Delay	0 to 1000 samp	Mid Delay	0 to 500 samp
Lo Mix	Off, -79.0 to 24.0 dB	Mid Mix	Off, -79.0 to 24.0 dB

Page 3

Hi Enable	On or Off		
Hi Drive	Off, -79.0 to 24.0 dB		
Hi Xfer1	-100 to 100%		
Hi Xfer2	-100 to 100%		
Hi Delay	0 to 500 samp		
Hi Mix	Off, -79.0 to 24.0 dB		

In/Out When set to “In” the effect is active; when set to “Out” the effect is bypassed.

Out Gain The overall gain or amplitude at the output of the effect.

CrossOver1 Adjusts one of the -6dB crossover points at which the input signal will be divided into the high, mid and low bands.

CrossOver2 Adjusts the other -6dB crossover points at which the input signal will be divided into the high, mid and low bands.

Enable Low, Mid, and High. Turns processing for each band on or off. Turning each of the 3 bands off results in a dry output signal.

Drive Low. Mid, and High. Adjusts the input into each transfer. Increasing the drive will increase the effects.

Xfer Low, Mid, and High; Xfer1 and Xfer2 for Mid and High. Adjusts the intensity of the transfer curves.

Delay Low, Mid, and High. Adjusts the number of samples the each signal is delayed.

Mix Low, Mid, and High. Adjusts the output gain of each band.

961 Tremolo
962 Tremolo BPM

A stereo tremolo or auto-balance effect

PAUs: 1

Tremolo and Tremolo BPM are 1 processing allocation unit (PAU) stereo tremolo effects. In the classical sense, a tremolo is the rapid repetition of a single note created by an instrument. Early music synthesists imitated this by using an LFO to modulate the amplitude of a tone. This is the same concept as amplitude modulation, except that a tremolo usually implies that the modulation rate is much slower.

Tremolo and Tremolo BPM provide six different LFO shapes (Figure 2), an additional shape modifier called “50% Weight”, “L/R Phase” for auto-balancing, and LFO metering. L/R Phase flips the LFO phase of the left channel for auto-balancing applications. The 50% Weight parameter bends the LFO shape up or down relative to it’s -6dB point (Figure 1). At 0dB, there is no change to the LFO shape. Positive values will bend the LFO up towards unity, while negative values will bend it down towards full attenuation. Additionally, LFO metering can be viewed on the bottom of PARAM2 page.

Tremolo also includes an LFO rate scale for AM synthesis, and Tremolo BPM provides tempo based LFO syncing including system syncing.

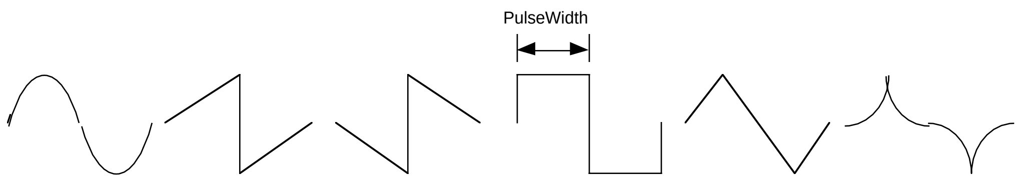


Figure 10-89 LFO Shapes available for Tremolo and Tremolo BPM

Parameters for Tremolo

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB

Page 2

LFO Rate	0 to 10.00 Hz	LFO Shape	Tri
Rate Scale	1 to 25088 x	PulseWidth	0 to 100 %
Depth	0 to 100 %	50% Weight	-6 to 3 dB
		L/R Phase	In or Out
A			
0% 50% 100%			

Parameters for Tremolo BPM

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
		Tempo	System, 0 to 255 BPM

Page 2

LFO Rate	0 to 12.00 x	LFO Shape	Tri
LFO Phase	0.0 to 360.0 deg	PulseWidth	0 to 100 %
Depth	0 to 100 %	50% Weight	-6 to 3 dB
		L/R Phase	In or Out
A			
0% 50% 100%			

In/Out When set to “In” the effect is active; when set to “Out” the effect is bypassed.

Out Gain The overall gain or amplitude at the output of the effect.

Tempo For Tremolo BPM. Basis for the rate of the LFO, as referenced to a musical tempo in BPM (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LFO Rate For Tremolo. The speed of the tremolo LFO in cycles per second.

LFO Rate For Tremolo BPM. The number of LFO cycles in one beat relative to the selected Tempo. For example, 1.00x means the LFO repeats once per beat; 2.00x twice per beat; etc...

Rate Scale For Tremolo. This multiplies the speed of the LFO rate into the audio range. When above 19x, the values increment in semitone steps. These steps are accurate when LFO Rate is set to 1.00 Hz.

LFO Phase For Tremolo BPM. This parameter shifts the phase of the tremolo LFO relative to an internal beat reference. It is most useful when Tempo is set to “System” and LFO Phase controls the phase of the LFO relative to MIDI clock.

Depth This controls the amount of attenuation applied when the LFO is at its deepest excursion point.

LFO Shape The waveform type for the LFO. Choices are Sine, Saw+, Saw-, Pulse, Tri, and Expon.

PulseWidth When the LFO Shape is set to Pulse, this parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to 50%. This parameter is active only when the Pulse waveform is selected.

50% Weight The relative amount of attenuation added when the LFO is at the -6dB point. This causes the LFO shape to bow up or down depending on whether this parameter is set positive or negative (Figure 1).

L/R Phase LFO phase relationship of the left channel. Flipping the left channel’s LFO out of phase causes the effect to become an auto-balancer.

963 AutoPanner

A stereo auto-panner

PAUs: 1

AutoPanner is a 1 processing allocation unit (PAU) stereo auto pan effect. The process of panning a stereo image consists of shrinking the image width of the input program then cyclically moving this smaller image from side to side while maintaining relative distances between program point sources (Figure 1). This effect provides six different LFO shapes (Figure 2), variable center attenuation, and a rate scaler that scales LFO rate into the audible range for a new flavor of amplitude modulation effects.

Final image placement can be monitored on the lower right of the PARAM2 page. The top meter labeled "L" shows the left edge of the image while the second meter labeled "R" shows the right edge. The entire image will fall between these two marks.

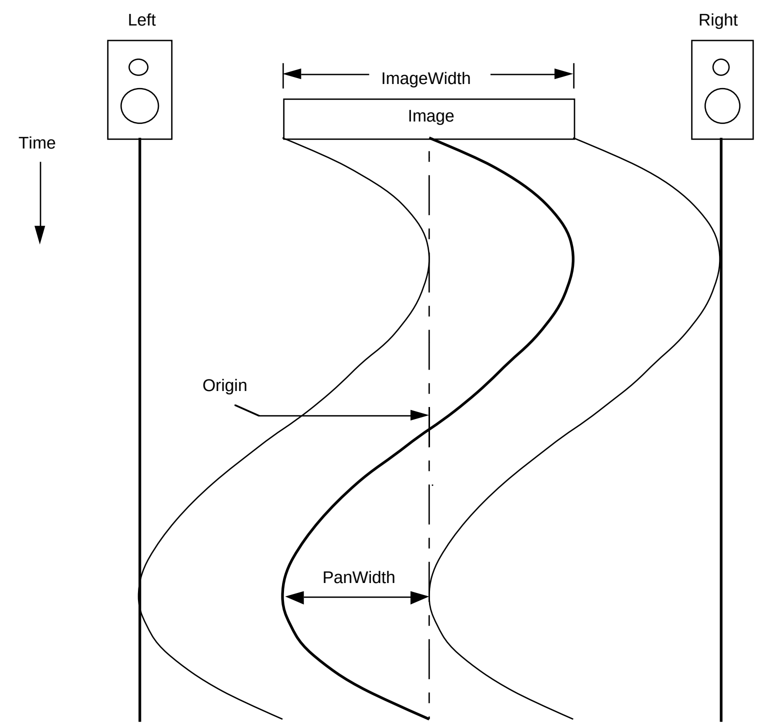


Figure 10-90 Stereo Autopanning

In Figure 10-90, ImageWidth is set to 50%, LFO Shape is set to Sine, Origin is set to 0%, and PanWidth is set to 100%

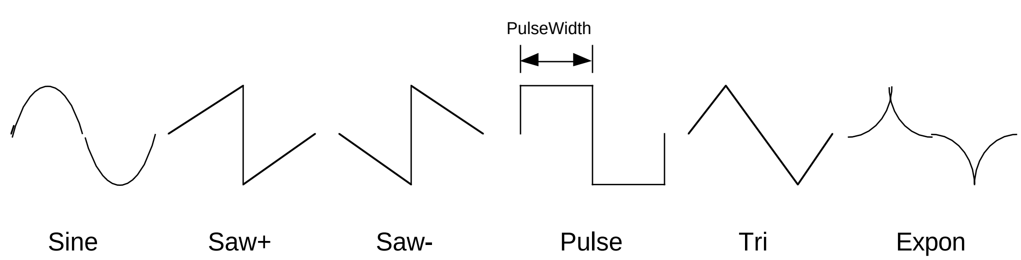


Figure 10-91 LFO Shapes available for AutoPanner

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB

Page 2

LFO Rate	0 to 10.00 Hz	LFO Shape	Tri
Rate Scale	1 to 25088 x	PulseWidth	0 to 100%
Origin	-100 to 100 %		
PanWidth	0 to 100 %	L	
ImageWidth	0 to 100 %	R	
CentrAtten	-12 to 0 dB	L C R	

In/Out When set to “In” the auto-panner is active; when set to “Out” auto-panner is bypassed.

Out Gain The overall gain or amplitude at the output of the effect.

LFO Rate The speed of the panning motion.

Rate Scale Multiplies the speed of the LFO rate into the audio range. When above 19x, the values increment in semitone steps. These steps are accurate when LFO Rate is set to 1.00 Hz.

Origin The axis for the panning motion. At 0%, panning excursion is centered between the listening speakers. Positive values shift the axis to the right, while negative values shift it to the left. At -100% or +100%, there is no room for panning excursion.

Pan Width The amount of auto pan excursion. This value represents the percentage of total panning motion available after Origin and ImageWidth are set.

ImageWidth The width of the original input program material before it is auto panned. At 0%, the input image is shrunk to a single point source allowing maximum panning excursion. At 100%, the original width is maintained leaving no room for panning excursion.

CentrAtten Amount the signal level is dropped as it is panned through the center of the listening stereo speaker array. For the smoothest tracking, a widely accepted subjective reference is -3dB. Values above -3dB will cause somewhat of a bump in level as an image passes through the center. Values below -3dB will cause a dip in level at the center.

LFO Shape The waveform type for the LFO. Choices are Sine, Saw+, Saw-, Pulse, Tri, and Expon.

PulseWidth When the LFO Shape is set to Pulse, this parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to 50%. This parameter is active only when the Pulse waveform is selected.

964 Dual AutoPanner

A dual mono auto-panner

PAUs: 2

Dual AutoPanner is a 2 processing allocation unit (PAU) dual mono auto pan effect. Left and right inputs are treated as two mono signals which can each be independently auto-panned. Parameters beginning with “L” control the left input channel, and parameters beginning with “R” control the right input channel. Autopanning a mono signal consists of choosing an axis offset, or Origin, as the center of LFO excursion, then adjusting the desired excursion amount, or PanWidth. Note that the PanWidth parameter is a percentage of the available excursion space after Origin is adjusted. If Origin is set to full left (-100%) or full right (100%) then there will be no room for LFO excursion. Control of six different LFO shapes (Figure 2), variable center attenuation, and a rate scaler that scales LFO rate into the audible range for a new flavor of amplitude modulation effects are also provided for each channel.

Final image placement can be seen on the bottom right of the PARAM2 and PARAM3 pages respectively for left and right input channels. The moving mark represents the location of each channel within the stereo field.

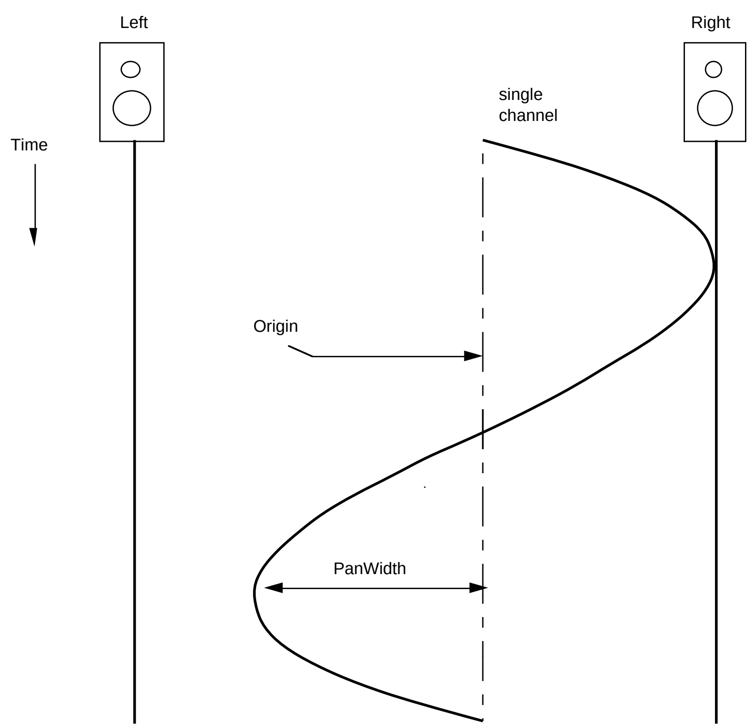


Figure 10-92 Mono autopanning

In Figure 10-92, LFO Shape is set to Sine, Origin is set to 15%, and PanWidth is set to 100%

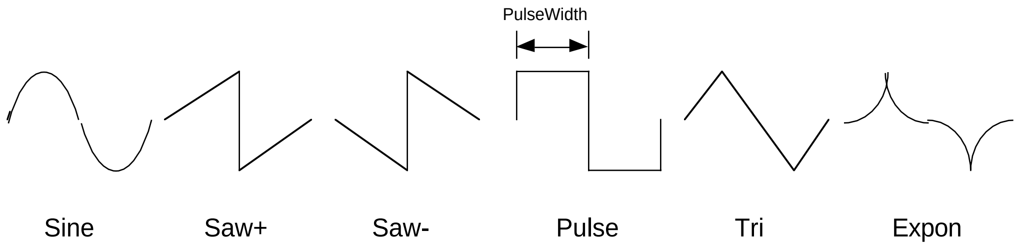


Figure 10-93 LFO Shapes available for Dual AutoPanner

Parameters

Page 1

L In/Out	In or Out	R In/Out	In or Out
L Out Gain	Off, -79.0 to 24.0 dB	R Out Gain	Off, -79.0 to 24.0 dB

Page 2

L LFO Rate	0 to 10.00 Hz	L LFO Shape	Tri
L RateScal	1 to 25088 x	L PlseWdth	0 to 100 %
L Origin	-100 to 100 %		
L PanWidth	0 to 100 %		
L CentrAtt	0 to 100 %	L	
		L C R	

Page 3

R LFO Rate	0 to 10.00 Hz	R LFO Shape	Tri
R RateScal	1 to 25088 x	R PlseWdth	0 to 100 %
R Origin	-100 to 100 %		
R PanWidth	0 to 100 %		
R CentrAtt	0 to 100 %	R	
		L C	

In/Out When set to “In” the auto-panner is active; when set to “Out” auto-panner is bypassed.

Out Gain The overall gain or amplitude at the output of the effect.

LFO Rate The speed of the panning motion.

Origin The axis for the panning motion. At 0%, panning excursion will be centered at the center of the listening speakers. Positive values shift the axis to the right, while negative values shift it to the left. At -100% or +100%, there is no room for panning excursion.

Pan Width The amount of auto pan excursion. This value represents the percentage of total panning motion available after Origin is set.

CentrAtten Amount the signal level is dropped as it is panned through the center of the listening stereo speaker array. For the smoothest tracking, a widely accepted subjective reference is

-3dB. Values above -3dB will cause somewhat of a bump in level as an image passes through the center. Values below -3dB will cause a dip in level at the center.

LFO Shape The waveform type for the LFO. Choices are Sine, Saw+, Saw-, Pulse, Tri, and Expon.

PulseWidth When the LFO Shape is set to Pulse, this parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to 50%. This parameter is active only when the Pulse waveform is selected.

965 SRS

Licenced Sound Retrieval System® or SRS™ effect

PAUs: 1

The SRS ™ algorithm has been licenced from SRS Labs, Inc. The following is from an SRS Labs press release:

SRS, the Sound Retrieval System, is based on the human hearing system. It produces a fully immersive, three-dimensional sound image from any audio source with two or more standard stereo speakers. Whether the signal is mono, stereo, surround sound or encoded with any other audio enhancement technology, SRS expands the material and creates a realistic, panoramic sound experience with no “sweet spot” or centered listening position. SRS is single-ended, requiring no encoding or decoding, and uses no artificial signal manipulation such as time delay or phase shift to produce its natural, true-to-life sound image.

The four SRS parameters control the ambience of the image, and may have different optimal settings depending on the amount of stereo content in the inputs. To match the optimal settings specified by SRS Labs, the bass and treble gains should be set to 0 dB. This algorithm will have no effect on mono signals.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Center	Off, -79.0 to 24.0 dB	Bass Gain	-79.0 to 24.0 dB
Space	Off, -79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB

In/Out When set to “In” the effect is active; when set to “Out” the effect is bypassed.

Out Gain The overall gain or amplitude at the output of the effect. Out Gain is not applied to the signal when the effect is bypassed.

Center The amount of “center channel” can be varied with this control.

Space The width of the image is controlled with this parameter.

Bass Gain The amount of ambience added to the Bass frequencies in the signals. A setting of 0 dB gives a best match to the optimizations of SRS Labs.

Treb Gain The amount of ambience added to the Treble frequencies in the signal. A setting of 0 dB gives a best match to the optimizations of SRS Labs.

966 Stereo Image

Stereo enhancement with stereo channel correlation metering

PAUs: 1

Stereo Image is a stereo enhancement algorithm with metering for stereo channel correlation. The stereo enhancement performs simple manipulations of the sum and difference of the left and right input channels to allow widening of the stereo field and increased sound field envelopment. After manipulating sum and difference signals, the signals are recombined (a sum and difference of the sum and difference) to produce final left and right output.

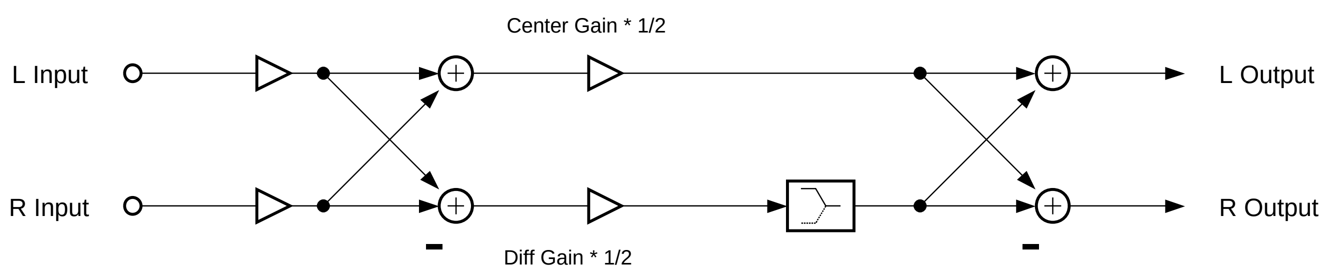


Figure 10-94 Block diagram of Stereo Image algorithm

The sum of left and right channels represents the mono or center mix of your stereo signal. The difference of left and right channels contains the part of the signal that contains stereo spatial information. The Stereo Image algorithm has controls to change the relative amounts of sum (or center) versus difference signals. By increasing the difference signal, you can broaden the stereo image. Be warned, though, that too much difference signal will make your stereo image sound “phasey”. With phasey stereo, acoustic images become difficult to localize and can sound like they are coming from all around or from within your head.

A bass shelf filter on the difference signal is also provided. By boosting only the low frequencies of the difference signal, you can greatly improve your sense of stereo envelopment without destroying your stereo sound field. Envelopment is the feeling of being surrounded by your acoustic environment. Localized stereo images still come from between your stereo loudspeakers, but there is an increased sense of being wrapped in the sound field.

The Stereo Image algorithm contains a stereo correlation meter. The stereo correlation meter tells you how alike or how different your output stereo channels are from each other. When the meter is at 100% correlation, then your signal is essentially mono. At 0% correlation, your left and right channels are the same, but polarity inverted (there is only difference signal). The correlation meter can give you an indication of how well a recording will mix to mono. The meter follows RMS signal levels (root-mean-square) and the RMS Settle parameter controls how responsive the meter is to changing signals. The ‘M’ part of RMS is “mean” or average of the squared signal. Since a mean over all time is neither practical or useful, we must calculate the mean over shorter periods of time. If the time is too short we are simply following the signal wave form, which is not helpful either, since the meter would constantly bounce around. The RMS Settle parameter provides a range of useful time scales.

See also the Stereo Analyze algorithm which allows you to experiment directly with sum and difference signals.

Parameters

Page 1

L In Gain	Off, -79.0 to 24.0 dB	R In Gain	Off, -79.0 to 24.0 dB
CenterGain	Off, -79.0 to 24.0 dB	Diff Gain	Off, -79.0 to 24.0 dB
L/R Delay	-500.0 to 500.0 samp	RMS Settle	0.0 to 300.0 dB/s

Page 2

DiffBassG	-79.0 to 24.0 dB		
DiffBassF	16 to 25088 Hz		
	Stereo Correlation		
	100 75 50 25 0%		

L In Gain The input gain of the left channel in decibels (dB).

R In Gain The input gain of the right channel in decibels (dB).

CenterGain The level of the sum of left and right channels in decibels (dB). The summed stereo signal represents the mono or center mix.

Diff Gain The level of the difference of left and right channels in decibels (dB). The difference signal contains the spatial component of the stereo signal.

L/R Delay If this parameter is positive, the left signal is delayed by the indicated amount. If it is negative, the right channel is delayed. You can use this parameter to try to improve cancellation of the difference signal if you suspect one channel is delayed with respect to the other.

RMS Settle Controls how fast the RMS meters can rise or fall with changing signal levels.

DiffBassG By boosting the low frequency components of the difference signal you can increase the sense of acoustic envelopment, the sense of being surrounded by an acoustic space. DiffBassG is the gain parameter of a bass shelf filter on the difference signal. DiffBassG sets how many decibels (dB) to boost or cut the low frequencies.

DiffBassF The transition frequency in Hertz (Hz) of the difference signal bass shelf filter is set by DiffBassF.

967 Mono -> Stereo

Stereo simulation from a mono input signal

PAUs: 1

Mono -> Stereo is an algorithms which creates a stereo signal from a mono input signal. The algorithm works by combining a number of band-splitting, panning and delay tricks. The In Select parameter lets you choose the left or right channel for you mono input, or you may choose to sum the left and right inputs.

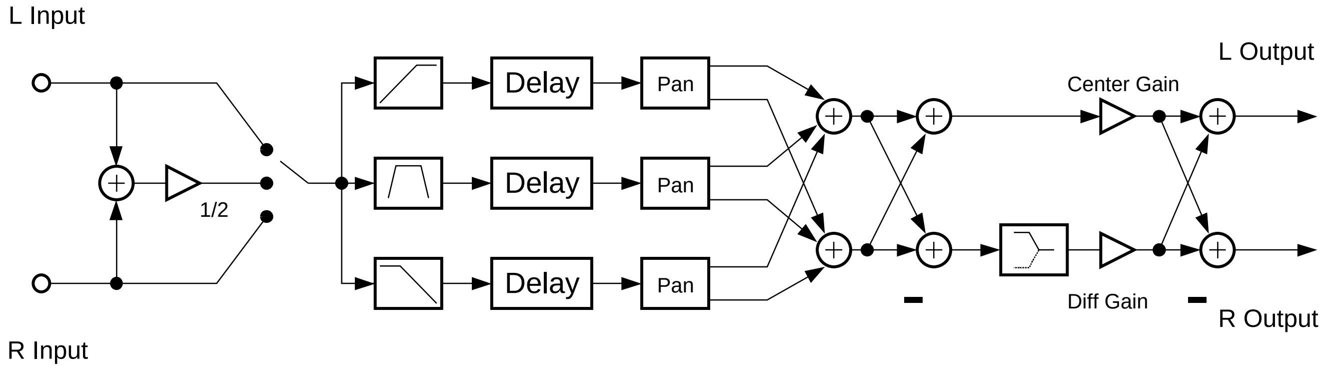


Figure 10-95 Block diagram of Mono -> Stereo effect.

The mono input signal is split into three frequency bands (Low, Mid, and High). The frequencies at which the bands get split are set with the Crossover parameters. Each band can then be delayed and panned to some position within your stereo field.

The final step manipulates the sum and difference signals of the pseudo-stereo signal created by recombining the split frequency bands. The sum of left and right channels represents the mono or center mix of your stereo signal. The difference of left and right channels contains the part of the signal that contains stereo spatial information. The Stereo Image algorithm has controls to change the relative amounts of sum (or center) versus difference signals. By increasing the difference signal, you can broaden the stereo image. Be warned, though, that too much difference signal will make your stereo image sound “phasey”. With phasey stereo, acoustic images become difficult to localize and can sound like they are coming from all around you or from within your head.

A bass shelf filter on the difference signal is also provided. By boosting only the low frequencies of the difference signal, you can greatly improve your sense of stereo envelopment without destroying your stereo sound field. Envelopment is the feeling of being surrounded by your acoustic environment. Localized stereo images still come from between your stereo loudspeakers, but there is an increased sense of being wrapped in the sound field.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
CenterGain	Off, -79.0 to 24.0 dB	Diff Gain	Off, -79.0 to 24.0 dB
In Select	L, R, or (L+R)/2	DiffBassG	-79.0 to 24.0 dB
		DiffBassF	16 to 25088 Hz

Page 2

Crossover1	16 to 25088 Hz		
Crossover2	16 to 25088 Hz		
Pan High	-100 to 100%	Delay High	0.0 to 1000.0 ms
Pan Mid	-100 to 100%	Delay Mid	0.0 to 1000.0 ms
Pan Low	-100 to 100%	Delay Low	0.0 to 1000.0 ms

In/Out The algorithm is functioning when In/Out is set to "In". If set to "Out, whatever is on the input channels gets passed to the output unaltered.

Out Gain The output gain of the pseudo-stereo signal in decibels (dB).

CenterGain The level of the sum of the intermediate left and right stereo channels in decibels (dB). The summed stereo signal represents the mono or center mix.

Diff Gain The level of the difference of the intermediate left and right stereo channels in decibels (dB). The difference signal contains the spatial component of the stereo signal.

In Select The input signal may come from the left L or right R input channel, or the left and right channels may be summed to obtain the mono signal (L+R)/2. You should set this parameter to match your Studio configuration.

DiffBassG By boosting the low frequency components of the difference signal of the intermediate stereo result, you can increase the sense of acoustic envelopment, the sense of being surrounded by an acoustic space. DiffBassG is the gain parameter of a bass shelf filter on the difference signal. DiffBassG sets how many decibels (dB) to boost or cut the low frequencies.

DiffBassF The transition frequency in Hertz (Hz) of the difference signal bass shelf filter is set by DiffBassF.

Crossover*N* The two Crossover parameters set the frequencies at which the band-split filters split the mono signal into three bands. The two parameters are interchangeable: either may have a higher frequency than the other.

Pan Low, Mid, and High. The panning of each band is separately controllable. -100% is fully left and 100% is fully right.

Delay Low, Mid, and High. The delays are set in milliseconds (ms).

968 Graphic EQ
969 Dual Graphic EQ

Dual mono 10 band graphic equalizer

PAUs: 3

The graphic equalizer is available as stereo (linked parameters for left and right) or dual mono (independent controls for left and right). The graphic equalizer has ten bandpass filters per channel. For each band the gain may be adjusted from -12 dB to +24 dB. The frequency response of all the bands is shown in the Figure 1. The dual graphic equalizer has a separate set of controls for the two mono channels (see Stereo Graphic Equalizer).

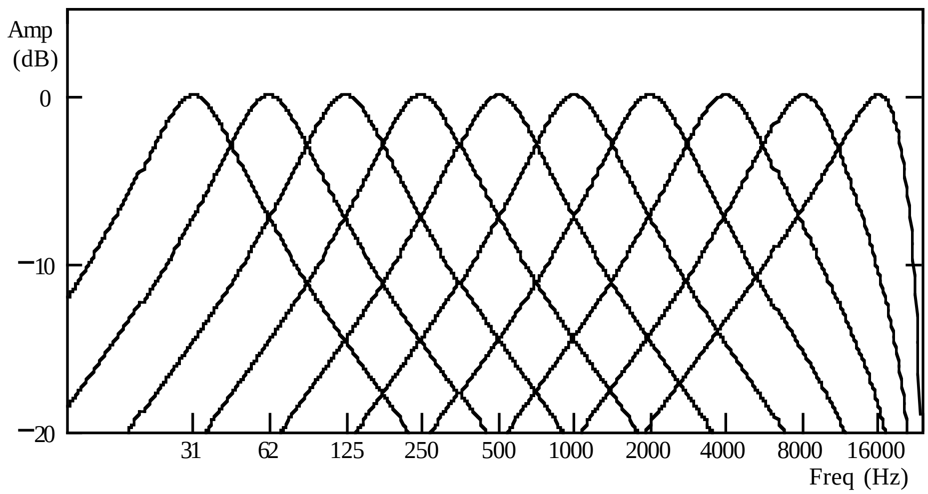


Figure 10-96 Filter Response of Each Bandpass Filter

Like all graphic equalizers, the filter response is not perfectly flat when all gains are set to the same level (except at 0 dB), but rather has ripple from band to band (see Figure 2). To minimize the EQ ripple, you should attempt to center the overall settings around 0 dB.

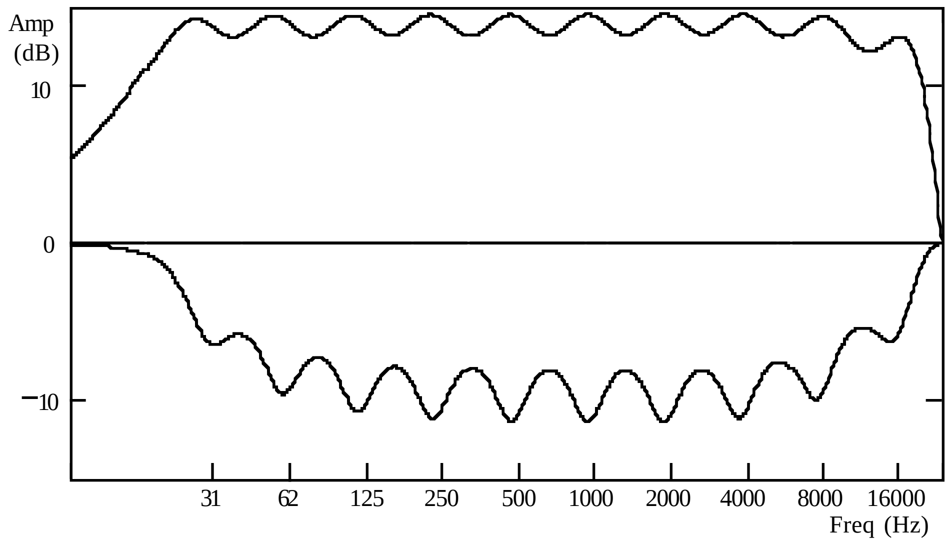


Figure 10-97 Overall Response with All Gains Set to +12 dB, 0 dB and -6 dB

Parameters for Graphic EQ

Page 1

In/Out	In or Out		

Page 2

31Hz G	-12.0 to 24.0dB	1000Hz G	-12.0 to 24.0dB
62Hz G	-12.0 to 24.0dB	2000Hz G	-12.0 to 24.0dB
125Hz G	-12.0 to 24.0dB	4000Hz G	-12.0 to 24.0dB
250Hz G	-12.0 to 24.0dB	8000Hz G	-12.0 to 24.0dB
500Hz G	-12.0 to 24.0dB	16000Hz G	-12.0 to 24.0dB

Parameters for Dual Graphic EQ

Page 1

L In/Out	In or Out	R In/Out	In or Out

Page 2

L 31Hz G	-12.0 to 24.0dB	L 1000Hz G	-12.0 to 24.0dB
L 62Hz G	-12.0 to 24.0dB	L 2000Hz G	-12.0 to 24.0dB
L 125Hz G	-12.0 to 24.0dB	L 4000Hz G	-12.0 to 24.0dB
L 250Hz G	-12.0 to 24.0dB	L 8000Hz G	-12.0 to 24.0dB
L 500Hz G	-12.0 to 24.0dB	L16000Hz G	-12.0 to 24.0dB

Page 3

R 31Hz G	-12.0 to 24.0dB	R 1000Hz G	-12.0 to 24.0dB
R 62Hz G	-12.0 to 24.0dB	R 2000Hz G	-12.0 to 24.0dB
R 125Hz G	-12.0 to 24.0dB	R 4000Hz G	-12.0 to 24.0dB
R 250Hz G	-12.0 to 24.0dB	R 8000Hz G	-12.0 to 24.0dB
R 500Hz G	-12.0 to 24.0dB	R16000Hz G	-12.0 to 24.0dB

In/Out When set to In the left channel equalizer is active; when set to Out the left channel equalizer is bypassed.

31Hz G Gain of the left 31 Hz band in dB.

62Hz G Gain of the left 62 Hz band in dB.

125Hz G Gain of the left 125 Hz band in dB.

250Hz G Gain of the left 250 Hz band in dB.

500Hz G Gain of the left 500 Hz band in dB.

1000Hz G Gain of the left 1000 Hz band in dB.

2000Hz G Gain of the left 2000 Hz band in dB.

4000Hz G Gain of the left 4000 Hz band in dB.

8000Hz G Gain of the left 8000 Hz band in dB.

16000Hz G Gain of the left 16000 Hz band in dB.

970 5 Band EQ

Stereo bass and treble shelving filters and 3 parametric EQs

PAUs: 3

This algorithm is a stereo 5 band equalizer with 3 bands of parametric EQ and with bass and treble tone controls. The user has control over the gain, frequency and bandwidth of each band of parametric EQ and control of the gain and frequencies of the bass and treble tone controls. The controls for the two stereo channels are ganged.

Parameters

Page 1

In/Out	In or Out		
Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	16 to 25088 Hz	Treb Freq	16 to 25088 Hz

Page 2

Mid1 Gain	-79.0 to 24.0 dB	Mid2 Gain	-79.0 to 24.0 dB
Mid1 Freq	16 to 25088 Hz	Mid2 Freq	16 to 25088 Hz
Mid1 Width	0.010 to 5.000 oct	Mid2 Width	0.010 to 5.000 oct

Page 3

Mid3 Gain	-79.0 to 24.0 dB		
Mid3 Freq	16 to 25088 Hz		
Mid3 Width	0.010 to 5.000 oct		

In/Out When set to “In” the tone controls are active; when set to “Out” the tone controls are bypassed.

Bass Gain The amount of boost or cut that the filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

Bass Freq The center frequency of the bass shelving filters in intervals of one semitone.

Treb Gain The amount of boost or cut that the filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

Treb Freq The center frequency of the treble shelving filters in intervals of one semitone.

Mid*n* Gain The amount of boost or cut that the filter should apply in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

Mid*n* Freq The center frequency of the EQ in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

Mid*n* Width The bandwidth of the EQ may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

998 FXMod Diagnostic

FXMod source metering utility algorithm

PAUs: 1

The FXMod diagnostic algorithm is used to obtain a metered display of FXMod sources. This algorithm allows you to view the current levels of any data sliders, MIDI controls, switches, or internally generated V.A.S.T. LFOs, ASRs, FUNs, etc. which are available as modulation sources. This algorithm has no effect on any signal being routed through it.

Up to eight modulation sources may be monitored simultaneously. Meters #1 through #4 can monitor bipolar sources, meaning sources which can have both positive and negative values. The range of the bipolar meters is -1 to +1. Four monopolar meters #5 through #8 provide better resolution, but the range is limited to 0 though +1. Use the monopolar meters for sources which you do not expect to go negative.

Eight parameters are provided to connect modulation sources to the meters. The parameter values are fixed at “NoDpth” and have no function except to connect sources to meters. To use the algorithm, save a Multieffect and Studio containing the algorithm, then go to one of the FXMod pages of your Program or Setup (with the Studio selected). Select the FX bus which contains the Multieffect using the FXMod Diagnostic algorithm, and choose one of the meter parameters (Bipole N or Monopole N). You will not be able to modify the Adjust or Depth fields, but you can select any source you want. Finally press the Edit button to re-enter the Studio and Multieffect editor where you can view the meters on parameter page 2.

Parameters

Page 1

Bipole 1	NoDpth	Monopole 5	NoDpth
Bipole 2	NoDpth	Monopole 6	NoDpth
Bipole 3	NoDpth	Monopole 7	NoDpth
Bipole 4	NoDpth	Monopole 8	NoDpth

Page 2

1	5
2	6
-1 0 1	0 0.5 1
3	7
4	8

Bipole *n* Use the Bipole parameters to attach bipolar modulation sources (can go positive or negative) to the bipolar meters. The parameters are not adjustable.

Monopole *n* Use the Monopole parameters to attach monopolar modulation sources (can go positive only) to the monopolar meters. The parameters are not adjustable.

999 Stereo Analyze

Signal metering and channel summation utility algorithm

PAUs: 1

Stereo Analyze is a utility algorithm which provides metering of stereo signals as its primary function. In addition to metering, the gains of the two channels are separately controllable, either channel may be inverted, and sum and differences to the two channels may be metered and monitored. If you use this algorithm with Live Mode, you can obtain a significant amount of information not only about your own mix, but of any recording you have in your library.

There are separate meters for the left and right output channels. Two types of meters are provided: peak and RMS. Meter display units are decibels relative to digital full scale (dBFS). The peak meters display the levels of the maximum signal peak that occurred during the meter update period (every 40ms). The RMS meter displays the average power of the input signal. RMS is an abbreviation for root-mean-square, so the signal is squared, averaged and a square root is taken. For a real-time meter, we do not take an average over all time, but rather average past signals with a stronger weighting to signals in the recent past than the far past. The RMS Settle parameter controls how strong the weighting is for recent signals over much older signals. RMS Settle is expressed in units of dB/s (decibels per second), meaning how fast the RMS meter can rise or fall with changing signal levels.

You can choose to meter and monitor normal left (L) and right (R) stereo signals, or with the Out Mode parameters, you can select normalized sum and differences of the left and right channels. The Out Mode parameters control the signals being passed to the outputs and to the meters: what you see on the meters are the signals to which you are listening. The Invert parameters provide a quick polarity reversal to the input signals. This polarity reversal occurs before sum and differences. The Invert parameters are actually redundant since Out Mode provides signal inversions as well. The left and right Out Mode parameters may be set to any of the following:

L	left channel
R	right channel
(L+R)/2	normalized sum of left and right
(L-R)/2	normalized difference of left minus right
-L	polarity reversed left channel
-R	polarity reversed right channel
-(L+R)/2	polarity reversed and normalized sum of left and right
(R-L)/2	normailized difference of right minus left

You may well ask why you would want to meter or monitor reversals or sums or differences of your stereo channels. One important case is to determine if your final mix is mono compatible -- very important if your mix is ever going to be broadcast on radio or television. Set both the left and right Out Mode parameters to (L+R)/2 to listen to the mono signal. If you find that parts of your mix disappear or start to sound metallic (comb filtered), you may have to go back and do some work on your mix.

The difference signal (L-R)/2 provides a measure of the stereo content of your mix and can be very indicative of mixing style. Listening to the difference signal of someone else's recordings can often demonstrate interesting techniques (and mistakes!) in stereo production. The difference signal contains everything that doesn't make it into the mono mix. Out of phase signals will appear only in the difference signal. Panned signals will appear in both the sum and difference signals to varying degrees. A delay between left and right channels will sound metallic (comb filtered or flanged) in both the sum and difference channels. If the entire mix seems to have a relative left/right delay, you can use the L/R Delay parameter to attempt to correct the problem. Positive delays are delaying the left channel, while negative delays are delaying the right channel.

By inverting one channel with respect to the other, you can hear what is characterised as “phasey-ness”. Usually in stereo recordings, you can localize the phantom image of sound sources somewhere between the two loudspeakers. With a phasey signal, the localization cue get mixed up and you may hear the sound coming from everywhere or within your head. Polarity reversals are provided in this algorithm so you can test for mistakes, or simply for experimentation.

Parameters

Page 1

L In Gain	Off, -79.0 to 24.0 dB	R In Gain	Off, -79.0 to 24.0 dB
L Invert	In or Out	R Invert	In or Out
L Out Mode	L	R Out Mode	R
L/R Delay	-500.0 to 500.0 samp	RMS Settle	0.0 to 300.0 dB/s

Page 2

Peak (-dBFS)			
L		R	
55 40 * 16 8 4 0		55 40 * 16 8 4 0	
L		R	
RMS (-dBFS)			

L In Gain The input gain of the left channel in decibels (dB).

R In Gain The input gain of the right channel in decibels (dB).

L Invert When set to on, the polarity of the left channel is reversed.

R Invert When set to on, the polarity of the right channel is reversed.

L Out Mode Determines which signal is to be metered (left meter) and passed to the left output. Choices are “L” (left), “R” (right), “(L+R)/2” (normalized sum), “(L-R)/2” (normalized difference), and polarity inverted versions of these.

R Out Mode Determines which signal is to be metered (right meter) and passed to the right output. Choices are “L” (left), “R” (right), “(L+R)/2” (normalized sum), “(L-R)/2” (normalized difference), and polarity inverted versions of these.

L/R Delay If this parameter is positive, the left signal is delayed by the indicated amount. If it is negative, the right channel is delayed. You can use this parameter to try to improve cancellation of the difference signal if you suspect one channel is delayed with respect to the other.

RMS Settle RMS Settle controls how fast the RMS meters can rise or fall with changing signal levels. Units are decibels per second (dB/s).

Chapter 11
Glossary

Algorithm In the K2600, a preset configuration of programmable digital signal processing functions. Each of a program's layers uses its own algorithm, which determines the type of synthesis each layer uses to generate its sound. FX presets also use algorithms, which determine what kind of DSP gets applied to the signal as it passes through a studio.

Aliasing A type of distortion that occurs in digitally sampled sounds when higher pitches (increased sample playback rates) introduce partials that were not present in the original sound. These partials may or may not be musically useful.

Amplitude The intensity of a signal, perceived as loudness in the case of audio signals.

Analog A term used widely in electronics-related fields to describe a method of representing information, in which the method of representation resembles the information itself. Analog synthesizers, for example, use gradual variations in electrical voltage to create and modify sounds. The oscillations in voltage are analogous to the waveforms of the sounds they generate. Compare Digital.

Bandwidth In terms of sound generation, the range of frequencies within which a device functions. The human ear has a "bandwidth" of almost 20 KHz (it can distinguish sound at frequencies from 20 Hz to 20KHz). The K2600's 20KHz bandwidth enables it to produce sounds that span the entire range of humanly audible sound.

Bank There are two types of banks in the K2600's memory: memory banks, which store and organize the programs and other objects you create, and Quick Access banks, where you can store programs and setups for one-button access while in Quick Access mode.

Cent 1/100th of a semitone. The standard increment for fine adjustment of pitch.

Continuous control A device that converts motion into a range of 128 possible values that can modulate a sound source. The Mod Wheel, a standard volume pedal, and controllers like Breath and Aftertouch are continuous controls. Compare Switch controls.

Control Source Anything that can be used to modify some aspect of a program's sound. LFOs, envelopes, Mod Wheel messages (MIDI 01), and FUNs are just a few examples of the K2600's control sources.

DSP Digital signal processing (see).

DSP Functions The K2600's collection of digital signal processing functions are what give the Variable Architecture Synthesis system its flexibility. Within each layer's algorithm, you can select from a long list of DSP functions like filters, EQ, oscillators, and a few that are unique to the K2600. Each DSP function has a corresponding page that enables you to assign numerous control sources to define how the DSP functions affect the sound of the program you're editing.

Default The starting condition of a system. The settings for the K2600's parameters are at their defaults when you unpack it, and they stay there until you change them. A hard reset will erase RAM and restore all parameters to their defaults.

Dialog A page that prompts you to enter information that the K2600 needs in order to execute an operation. Dialogs appear, for example, when you initiate a Save or Delete operation.

Digital A term used widely in electronics-related fields to describe a method of representing information as a series of binary digits (bits)—1s and 0s. Digital computers process these strings of 1s and 0s by converting them into an electrical signal that is always in one of two very definite states: “on” or “off.” This is much more precise than the analog method, therefore digital computers can operate at speeds unattainable by analog devices. Digital synthesizers like the K2600 are actually computers that process vast strings of digital information signals, eventually converting them (at the audio output) into the analog signals that flow into PAs and other audio systems. See also Analog.

Digital Signal Processing The term “Signal processing” refers to a vast range of functions, all of which have in common the fact that they act upon an electric current as it flows through a circuit or group of circuits. A simple form of signal processing is the distortion box used by many guitarists. *Digital* signal processing refers to similar processes that are performed by digital (see) circuitry as opposed to analog (see) circuitry. Many of the effects devices available today use digital signal processing techniques.

Drum Program Any program consisting of more than three layers. So called because in the K2000, a special channel was required to handle programs with more than three layers—which typically were-multi-timbral percussion programs.

Editor The complete set of parameters used to modify a particular aspect of the K2600, for example, the currently selected Program, which is modified with the Program Editor. The Program Editor spans several display pages, which can be viewed by using the soft buttons (the ones labeled **<more>**.

Envelope An aperiodic modifier. In other words, a way to cause a sound to change over time without repeating the change (unlike periodic modifiers like LFOs, which repeat at regular intervals).

File A group of objects stored to a floppy or hard disk, or loaded into the K2600’s RAM from disk.

Global In this manual, used primarily in reference to control sources. A global control source affects all notes in a layer uniformly. If a layer uses a global control source, that control source begins to run as soon as the program containing it is selected. Its effect on each note will be completely in phase, regardless how many notes are being played. Compare Local.

Hard Reset Resets all parameter values to their defaults, and completely erases the contents of RAM. Press the Reset button in Master mode to do a hard reset. This is a quick way to restore the factory defaults to your K2600, but *everything* in RAM (all the objects you’ve created) will be erased, so objects you wish to keep should be saved to disk or SyxEx dump. A hard reset should not be used to recover if your K2600 is hung up, except as a last resort. See Soft Reset.

KB3 Program Uses oscillators to emulate tone wheel organs. Doesn’t use VAST processing; no layers, keymaps, or algorithms. Requires a special channel called the KB3 channel.

Keymap A keymap is a collection of samples assigned to specific notes and attack velocities. Keymaps usually contain numerous sample roots pitch-shifted across a range of several notes. When you trigger a note, the keymap tells the K2600 what sound to play, at what pitch, and at what loudness.

LFO	Low frequency oscillator. An oscillator is an electrical signal that cycles regularly between a minimum and maximum amplitude. The simplest oscillating waveform is the sine wave, but an LFO waveform can have almost any shape. The number of times each second that an oscillator repeats itself is called its frequency, which is measured in Hertz (Hz). Anything up to 50 Hz is considered low-frequency in musical applications. Use an LFO whenever you want to generate a *periodic* (repeating) effect. Adjusting the rate of the LFO will change the repetition rate of the effect.
Layer	A layer consists of a keymap processed through an algorithm. Layers can be stacked together within a program. Each layer uses one of the K2600's 48 available voices. Each K2600 program can contain up to three layers—except drum channel programs, which can contain up to 32 layers.
Leslie effect	This classic vibrato effect was originally created by mounting a speaker in its cabinet so the speaker could be rotated at varying speeds. This applied a vibrato of varying rate to all sounds played through the rotating speaker.
Local	In this manual, used primarily in reference to control sources. A local control source affects each note in a layer independently. For example, if a local LFO is used as a control source, a separate LFO cycle will begin with each note start. The LFOs don't run in phase unless notes are started simultaneously. Compare Global.
Memory banks	The K2600's memory is divided into ten spaces where you can store any object you edit. These spaces are called banks. Each bank can hold up to 100 objects of each type, so we refer to them as the 100s bank, the 200s bank, and so on. The ID of an object determines which bank it's stored in. An object with an ID of 399, for example, would be stored in the 300s bank. ROM objects are stored in the Zeros and 100s banks. RAM objects can be stored in any bank.
MIDI	Musical Instrument Digital Interface. A specialized format for representing musical information in terms of standardized computer data, which enables electronic musical instruments to communicate with computers
MIDI device	Any device—keyboard, computer, wind instrument, etc.—that is capable of transmitting and receiving MIDI messages. Also known as a MIDI controller, or a MIDI source.
MIDI Master	A MIDI device that is configured to control one or more other MIDI devices. The MIDI Out port of the master is connected by cable to the MIDI In port(s) of the slave device(s).
MIDI Slave	A MIDI device that is configured to receive MIDI messages from a master device. The MIDI In port of the slave is connected by cable to the MIDI Out port of the master.
Nonlinear DSP Function	Without getting technical, nonlinear DSP functions like SHAPER and WRAP add waveforms to those already present in a sound, while linear DSP functions act upon the existing waveforms without adding new ones.
Note State	Any K2600 note is either on or off; this is its note state. Normally, any given note's Note State switches on when you strike the key for that note. It switches off when you release the key, and any sustain controls you may have applied to the note (Sustain or Sostenuto pedal, etc.). Also see the index entry for Note State.
Object	A chunk of information stored in the K2600's memory. Programs, setups, keymaps, and samples are all objects. There are several others as well. Also see the index entry for Objects.

Page A set of performance or programming parameters that appear as a group in the display. The entry level page for each mode appears when you select the mode. Most other pages are selected with the soft buttons, from within an editor.

Parameter A programming feature. The name of the parameter describes the function it controls—transposition, for example. Each parameter has a value associated with it, which indicates the status of the parameter.

Pixel A contraction of "picture element." The K2600's display consists of a screen with small square dots (the pixels). Each pixel lets light through or blocks it depending on whether it is receiving an electrical charge. The combination of light and dark dots creates a pattern that you recognize as text or graphics. The K2600's display is 240-by-64 pixels, in other words, 64 horizontal rows, each containing 240 pixels, for a total of 15360 pixels.

Program The K2600's basic performance-level sound object. Programs can consist of up to 3 layers (32 layers for drum programs); each layer has its own keymap (set of samples) and sound-processing algorithm.

Program Editor The set of parameters that lets you modify the sound of ROM or RAM programs. Enter the Program Editor by pressing the EDIT button while in Program mode, or any time the currently selected parameter has a program as its value.

RAM Random Access Memory, one of the two basic types of computer memory. RAM can be both read from and written to. When you load samples into the K2600, or save a program you've created, you're writing to RAM. Compare ROM.

ROM Read Only Memory, one of the two basic types of computer memory. You can retrieve the information stored in ROM, but you can't write (save) new information to it. The onboard sounds of your K2600 are stored in ROM.

Sample A digital recording of a sound that can be assigned to a keymap as part of the process of building a program. Samples are stored in ROM (factory-installed) or in RAM (loaded from disk).

SCSI Pronounced "scuzzy," this acronym stands for Small Computer Systems Interface. It's simply a standardized form of information exchange that allows any SCSI equipped device to communicate with any other SCSI device. Two or more SCSI devices—they can be computers, hard disks, printers, just about anything that sends or receives information in standardized form—are connected via special cables to their SCSI ports. This configuration is much faster than serial information exchange, the precursor to SCSI.

SMDI Pronounced "smiddy," this acronym stands for SCSI Musical Data Interchange. It's a new format for data transfer, based on the SCSI format, which uses parallel input/output rather than serial, as used by MIDI and standard SCSI operations. This enables data to flow much faster. You can use SMDI to transfer samples to and from the K2600 using software packages from Passport and Opcode.

SMF Standard MIDI File. MIDI Type 0 files are single track, while MIDI Type 1 files are multi-track. The K2600 can read and write Type 0 files and read Type 1 files.

Semitone In "Western" music, the standard interval between the twelve notes in the scale. There are twelve semitones to an octave. The interval between C and C$^{\#}$ is one semitone.

Setup A multi-timbral performance object. A setup consists of three zones, each of which can be assigned its own program, MIDI channel, and control assignments. These assignments control the K2600's operation while in Setup mode, as well as determining the Program Change numbers and controller messages the K2600 sends via MIDI.

Soft Reset Returns the K2600 to Program mode without affecting the contents of RAM. Press the **+/-**, **0**, and **Clear** buttons to do a soft reset. If your K2600 is hung up for some reason, this will usually get take care of the problem. See Hard Reset.

Switch control A device that converts motion into discrete on/off signals. A switch control, like the Sustain pedal, is either on or off. Compare continuous control.

Toggle As a verb, to switch between (usually) two conditions using a device that makes the switch. As a noun, the device that makes the switch. For example, pressing the **View** soft button on the top level Program-mode page toggles between small-type and large-type views of the current program.

Value The current setting of a parameter. Each parameter has a range of available values, one of which you select while editing. The Transposition parameter on the Program-mode page, for example, has a default value of **0**. Change the value to change the parameter's effect on the current program.

VAST Variable Architecture Synthesis Technology. The term created by Kurzweil engineers to describe the multi-faceted capabilities of the K2600, combining sample playback (ROM and RAM), and waveform generation with a broad array of processing functions. This architecture provides preset algorithms created by Kurzweil sound engineers, which include filters, distortion, panning, EQ, waveform oscillators, waveform shaper, hard sync oscillators, amplitude modulation, gain, crossfade, and more. VAST is a registered trademark of Young Chang Akki Co. Ltd.

Zero Crossing Any of a number points in the digital representation of a sound's waveform where the digital signal is neither positive or negative. When looping samples, starting the loop at one of these points will reduce or eliminate the click or change in timbre that can occur in sample loops.

Appendix A Specifications

K2600 Features

- 240 x 64-pixel backlit fluorescent graphic display with adjustable contrast
- 3.5-inch floppy disk drive, for DD or HD disks, DOS compatible
- MIDI In, Thru, and Out with selectable second MIDI Out
- MIDI LED to indicate MIDI activity

- 48-note polyphony with dynamic voice allocation
- Multi-timbral, for multi-track sequencing and recording
- Hundreds of factory preset programs, and mroe than 100 factory preset setups
- Up to three layers per program, up to 32 layers for drum programs
- Receives mono (channel) pressure and poly (key) pressure
- Eight-zone setups transmit on eight MIDI channels with independent programmable controls
- Fully featured onboard sequencer for recording from keyboard or via MIDI; loads and plays MIDI Type 0 sequences
- Easy-to-use programming interface including soft buttons, Alpha Wheel, and alphanumeric pad

- Eight Megabytes of 16-bit sample ROM, including acoustic instrumental sounds, waveforms, and noise
- 20 KHz maximum bandwidth
- Optional stereo sampler with analog and digital inputs
- AES/EBU I/Os and optical I/O
- Sound ROM expandable to a total of 28 Megabytes
- Two SIMM sockets for optional sample RAM—up to 128 Megabytes
- Stereo sample playback capability
- Akai® S1000, Roland,® and EPS® sample disk compatibility

- Two 1/4-inch mixed audio outputs (stereo pair)
- Eight 1/4-inch audio outputs programmable as four stereo pairs or as eight separate outputs, with insert capability for effects patching
- Stereo headphone jack

- 500K battery-backed RAM for user programs, setups and other objects, expandable to 1500K
- Two SCSI ports for connection with external SCSI disks, CD-ROM drives, or personal computers
- Optional internal hard disk
- Optional eight-channel interface to AES, ADAT, DA-88

- Realtime DSP for each voice: 31 programmable DSP algorithms incorporating filters, EQ, distortion, panning, pulse width modulation, and more; up to 3 programmable DSP functions per voice
- Filters: Lowpass, Highpass, Allpass, Bandpass, Notch, programmable resonance
- Programmable stereo multi-effects on MIX outputs, including simultaneous reverb, chorus, delay, flanging, EQ—and more
- Realtime internal and MIDI control of effects parameters

- MIDI standard sample dump/load capability
- SMDI sample dump/load capability
- System Exclusive implementation
- MIDIScope™ for analyzing MIDI events

Environmental Specifications

Temperature Ranges

For operation:	minimum	41° F (5° C)
	maximum	104° F (40° C)
For storage:	minimum	-13 ° F (-25° C)
	maximum	186° F (85° C)

Relative Humidity Ranges (Non-condensing)

Operation and storage:	5—95%

Physical Specifications

Overall dimensions	K2600R		K2600		K2600X	
Width	16.9 in[1]	43.0 cm	47.8 in	121.4 cm	54.3 in	137.9 cm
Depth	13.9 in	35.4 cm	17.8 in	45.1 cm	17.8 in	45.1 cm
Height	5.1 in	13.0 cm	4.8 in	12.2 cm	4.8 in	12.2 cm
Weight	24.65 lb	11.2 kg	55.5 lb	25.2 kg	72.0 lb	32.7 kg

1. Excluding rack-mount brackets

Electrical Specifications

AC supply: selectable; 100V, 120V, 220V, or 240V. 1.0 amps at 120 volts nominal

Safe Voltage Ranges

Voltage setting:	100V	120V	220V	240V
Safe voltage range:	85—107	95—125	180—232	190—250
Safe frequency range:	48—65	48—65	48—65	48—65

If the voltage drops below the minimum safe level at any voltage setting, the K2600 will reset, but no data will be lost. If the voltage exceeds the maximum safe level, the K2600 may overheat.

Analog Audio Specifications

Audio Jacks

- 1/4-inch TRS balanced/unbalanced
- Tip = Positive
- Ring = Negative
- Sleeve = Chassis Ground

Separate Outputs

	Balanced	Unbalanced
Maximum Output	21 dBu	15 dBu
Output Impedance	200 Ω	200 Ω

Mix Outputs

	Balanced	Unbalanced
Maximum Output	27 dBu	21 dBu
Output Impedance	200 Ω	200 Ω

Headphone Output

Maximum Output	21 dBu
Output Impedance	47 Ω

MIDI Implementation Chart

Model: K2600

Manufacturer:
Young Chang

Date: 3/21/95
Version 1.0

Digital Synthesizers

Function		Transmitted	Recognized	Remarks
Basic Channel	Default	1	1	Memorized
	Changed	1 - 16	1 - 16	
Mode	Default	Mode 3	Mode 3	Use Multi mode for multi-timbral applications
	Messages			
	Altered			
Note Number			0 - 127	0–11 sets intonation key
	True Voice	0 - 127	0 - 127	
Velocity	Note ON	O	O	
	Note OFF	O	O	
After Touch	Keys	X	O	
	Channels	O	O	
Pitch Bender		O	O	
Control Change		O 0 - 31 32 - 63 (LSB) 64 - 127	O 0 - 31 32 - 63 (LSB) 64 - 127	Controller assignments are programmable
Program Change		O 1 - 999	O 1 - 999	Standard and custom formats
	True #	0 - 127	0 - 127	
System Exclusive		O	O*	
System Common	Song Pos.	O	O	
	Song Sel.	O	O	
	Tune	X	X	
System Real Time	Clock	O	O	
	Messages	O	O	
Aux Messages	Local Control	O	O	
	All Notes Off	O	O	
	Active Sense	X	X	
	Reset	X	X	
Notes		*Manufacturer’s ID = 07 Device ID: default = 0; programmable 0–127		

Mode 1: Omni On, Poly
Mode 3: Omni Off, Poly

Mode 2: Omni On, Mono
Mode 4: Omni Off, Mono

O = yes
X =no

Appendix B
SysEx Control of KDFX

Any KDFX parameter that can be set to a destination of FXMod can also be controlled by MIDI system exclusive (SysEx) messages. This takes a little more effort, but allows more flexibility. It's especially useful when the K2600 is in Master effects mode (the FX Mode parameter on the Effect-mode page is set to Master). It's also a way to get additional real-time control—beyond the 18 FXMods that are available for a given program or setup.

Note that using SysEx control temporarily disables FXMod control for the corresponding parameter. For example, if a studio's Mix level is controlled by an FXMod, then you send a SysEx message to change it, the FXMod that was controlling the Mix level is disabled, and won't take effect again until the program or setup containing the FXMod gets selected.

You'll find general information about the K2600's SysEx implementation in Chapter 7.

SysEx Message Structure

A standard SysEx message is a string of hexadecimal numerals, each of which represents a byte of MIDI data ranging in value from 0 to 127—for example 2A, which represents the decimal numeral 42: (2 x 16) + 10). The hexadecimal numerals correspond to particular SysEx commands. Many of these commands are standardized by the MIDI Specification. Others are assignable by individual manufacturers.

Every SysEx command consists of three basic parts: header, body, and end. The header includes general data, like where the message is intended to go, and what type of message it is. The body issues the specific commands you want to send, and the end simply indicates that the SysEx message is finished.

Header

The following table provides the header information required for sending a KDFX-control SysEx message to the K2600.

Hexadecimal Value	Corresponding Decimal Value	Corresponding SysEx Command
F0	240	Start of SysEx message
07	7	Manufacturer ID (7 is Kurzweil/Young Chang)
00	00	Unit ID; if you're sending SysEx from the same source to multiple K2600s, use a different ID value for each one
78	120	Product ID (78 is K2000/K2500/K2600)
1B	27	Message type (1B is KDFX control)

Every KDFX-control SysEx message you send to the K2600 must start with this string of numerals. This lets the K2600 know that the remainder of the message contains specific KDFX-control instructions.

Body

The body of each SysEx message is where you issue one or more specific commands for KDFX control. Each specific command consists of four bytes (a string of four hexadecimal numerals). Each SysEx message you send can contain as many of these specific commands as you want.

Command Type	Allowable Values (Hexadecimal)	Allowable Values (Decimal)	Description
Device selection	00–2E	0–46	Studio component to be controlled (FXBus1, for example)
Parameter selection	Depends on device value	Depends on device value	Parameter to be controlled (Mix Lvl, for example)
Parameter value: MSB	00, 01, 7F	0, 1, 127	With LSB, sets value of parameter to be controlled
Parameter value: LSB	00–7F	0–127	Combined with MSB, sets value of parameter to be controlled

Table B-1 SysEx Message Body

See *MSB and LSB* on page -4 for an explanation of how to use MSB and LSB to send values in the range from -128 to 255.

End

The last hexadecimal numeral in a SysEx message is always F7 (127 decimal), which indicates the end of the SysEx message.

Device Codes

These codes identify the studio component that you want to control via SysEx. Use one of these values for the device selection byte in the body of your SysEx message.

Device Code (Hexadecimal)	Device Code (Decimal)	Studio Component
00	0	Send1 for Input A (or for A Left if Input A receives a mono signal)
01	1	Send1 for Input A Right (if Input A receives a mono signal)
02	2	Send1 for Input B (or for B Left if Input B receives a mono signal)
03	3	Send1 for Input B Right (if Input B receives a mono signal)
04	4	Send1 for Input C (or for C Left if Input C receives a mono signal)
05	5	Send1 for Input C Right (if Input C receives a mono signal)
06	6	Send1 for Input D (or for D Left if Input D receives a mono signal)
07	7	Send1 for Input D Right (if Input D receives a mono signal)
08–0F	8–15	Send2 for Inputs A–D (if input is stereo, use 08, 0A, 0C, and 0E)
10–17	16–23	1st EQ block for Inputs A–D
18–1F	24–31	2nd EQ block for Inputs A–D
20, 22, 24, 26	32, 34, 36, 38	Aux send for FXBuses 1–4
21, 23, 25, 27	33, 35, 37, 39	Mix send for FXBuses 1–4
28	40	Mix send for Aux bus
29	41	Final mix
2A	42	FX Preset for Aux bus
2B–2E	43–46	FX Preset for FXBuses 1–4

Parameter Codes

These codes identify the specific parameters for each studio component (device). Use one of these values for the parameter selection byte in the body of your SysEx message.

Device Code (Hexadecimal)	Parameter Code (Hexadecimal)	Parameter Code (Decimal)	Parameter
00–0F	00	0	Level
	01	1	Pan or Balance
	02	2	Width (for stereo inputs only)
10–1F	00	0	Gain (or Frequency if EQ block is hipass or Lopass)
	01	1	Frequency
20–29	00	0	Level
	01	1	Balance
2A–2E	00	0	Wet/Dry (or In/Out)
	01–2B	1–43	Variable, depending on FX Preset

Here's an example, which sets a value of 50% for the Wet/Dry mix of the effect on the Aux bus. We've included both hexadecimal and decimal values.

	Start	Manufacturer ID	Unit ID	Product ID	Message Type	Device Selection	Parameter Selection	Value MSB	Value LSB	End
Hex	F0	07	00	78	1B	2A	00	00	32	F7
Dec	240	7	0	120	27	42	0	0	50	247

MSB and LSB

The K2600 can accept either unsigned (positive only) or signed (positive and negative) values. Unsigned values can range from 0 to 255, and signed values can range from -128 to 127. Both of these ranges require eight bits of MIDI information. Since each byte of MIDI information contains only 7 meaningful bits, you need two bytes to send eight bits of information. The K2600 interprets these bytes as a two-byte pair and not as unrelated bytes. The first byte, called the most-significant byte (MSB) sets the general range of the value, while the second byte (the least-significant byte or LSB) sets the specific range. The following table shows several decimal values and the corresponding MSB-LSB hexadecimal values.

Decimal Value	Corresponding Hexadecimal Value	Corresponding SysEx Command	
		MSB	LSB
255	00FF	01	7F
192	00C0	01	40
128	0080	01	00
127	007F	00	7F
64	0040	00	40
0	0000	00	00
-1	FFFF	7F	7F
-64	FFC0	7F	40
-127	FF81	7F	01
-128	FF80	7F	00

Here’s a different way to look at it:

Parameter Value (Decimal)	MSB (Hexadecimal)	LSB
Unsigned, 128 to 255	01	(Parameter Value - 128 decimal)
Unsigned, 0 to 127	00	Parameter Value (decimal)
Signed, 0 to 127	00	Parameter Value (decimal)
Signed, -128 to -1	7F	(Parameter Value + 128 decimal)

For example, if you wanted to send a value of 216, the MSB would be 01 hex, and the LSB would be (216 - 128), or 88 decimal (58 hex). To send a value of -32, the MSB would be 7F, and the LSB would be (-32 + 128), or 96 decimal (60 hex).

If you’re using a dedicated MIDI source to generate SysEx, you might not need to calculate the parameter values, since the MIDI source might do it for you. For example, with one well-known MIDI fader box, the following values configure a fader for control over the Wet/Dry mix of the effect on the Aux bus:

Function	**String**
String	**F0 07 00 78 1B 2A 00 pr pr F7**
Min	**0**
Max	**100**
Param Format	**2Byte**, **7Bits**, **hi -> lo**

Moving the fader changes the values represented by **pr**.

Appendix C
Standard K2600 ROM Objects

In This Appendix

The objects listed in this Appendix are current with operating system version 1.01. Your K2600 probably has version 1.01 objects installed. Here's how you can check the version of the objects you have installed:

1. Press the **Master** mode button to enter Master mode.
2. Select the Intonation parameter.
3. Change its value to **18**. You should see something like this: **18 Obj B1.00**. This "intonation table" is actually the version number of the K2600's basic object file (hence the **B**). If you scroll higher in the list, you may see other version numbers, depending on the ROM block options you have.

If your instrument doesn't have version 1.01 objects, you can get them from our website:

http://www.youngchang.com/kurzweil/html/downloads.html

K2600 Program List

The preset programs in the K2600 are organized by instrument category. You'll find a few representatives of each instrument sampled for the base ROM sound block, as well as synthesized instrument emulations, commonly used synthesizer timbres, and templates for new programming. We hope you find it a good starting point for your own work.

Setup List

The setup is a combination of up to eight zones, each with independent MIDI channel and control assignments. Setups can be played on rack models via the Local Keyboard Channel feature: in MIDI mode on the RECEIVE page, change the value of the LocalKbdCh parameter from **None** to a channel of your choice, and set your MIDI source to send on only that channel. Now, any note that comes in on that channel will be remapped according to the display channel (in Program mode) and according to the setup (in Setup mode).

Conventional Controller Assignments

There are 99 factory setups in the K2600. You'll find unique internal program combinations, arpeggiator examples, special ribbon and controller functions, and templates for user-defined setups. With as many as 24 assignable controllers shared among 8 independent zones, K2600 setups can be quite powerful, and they require some experimentation to find all their features and nuances. In order to make this process easier, many setups are programmed according to certain conventions. The sliders generally provide mixing capabilities either as group faders or individual zone faders. They also provide control over timbre, effects mix, and clock tempo. Other conventions include those listed in Table -C-1 (these are the default settings for Setup **97 ControlSetup**).

Slider A: Data	Continuous Controller Pedal 1: Foot (MIDI 4)
Slider B: MIDI 22	Continuous Controller Pedal 2: Breath (MIDI 2)
Slider C: MIDI 23	Small Ribbon Position: Aux Bend 2
Slider D: MIDI 24	Small Ribbon Pressure: Mono Pressure
Slider E: MIDI 25	Large Ribbon: Aux Bend 1
Slider F: MIDI 26	Pitch Wheel: BendUp
Slider G: MIDI 27	Mod Wheel: MWhl
Slider H: MIDI 28	Panel Switch 1: Arpeggiator On/Off
Footswitch 1: Sustain	Panel Switch 2: MIDI 29
Footswitch 2: Sostenuto	Mono Pressure: MPress
Footswitch 3: Soft Pedal	
Footswitch 4: TapTempo	

Table C-1 Default Physical Controller Assignments

MIDI notes can be triggered from many controllers including pedals, switches, sliders and the ribbons.

Special Purpose Setups

There are three special setups at the end of the Zeros bank:

97 Control Setup	Lets you define controller assignments in Program mode. You can customize and select the control setup on the MIDI-mode TRANSMIT page.
98 Clear Setup	A template for creating your own control assignments from a clear palette.
99 Default Setup	Lets you create your own setups from our common settings. The NewZn parameter uses this setup as its template for creating new zones.

Programs

1	Concert Piano 1
2	Stereo Solo Pno
3	Brt Concert Pno
4	Rok Piano
5	Piano for Layers
6	DrkPno^ArakisPno
7	Honky-Tonk
8	Pno&Syn/AcString
9	ClassicPiano&Vox
10	E Grand Stack
11	Piano D'Biggy
12	DynEPiano^EPnoPF
13	FM-ish E-Piano
14	Suitcase E Piano
15	PhunkEPno^FonkMW
16	Clavinators 1^2
17	Spaceychord 1^2
18	D'Nu Clavs 1^2^3
19	VAST B3! PW+CC2
20	Rock Organ
21	Ballad Organ
22	Gospel Organ
23	Overdrive Organ
24	Perc Organ
25	Chiffy Pipes
26	Offertory
27	Pedal Pipes
28	Church Organ
29	DualBass^SlapBas
30	Warm Bass 1^2
31	SustBass^MixBass
32	RickenBass
33	Synth Fretless
34	Moogy Bass 1^2
35	Moogy Bass 3^4
36	MatrixBigBass1^2
37	PittBass mono1^2
38	Punch Bass 1^2
39	Tee Bee This MW
40	BottomFeed^Pulse
41	SkoolBass^Simple
42	MonoBass^FixBass
43	Ravelike Basses
44	Junosis^Geo-Bass
45	AceBass^ChirpBas
46	Gritz
47	Homey Saw
48	OG
49	Lowdown Bass
50	SquashStudio Kit
51	Retro Skins MW
52	2 Live Kits 2 MW
53	Garage Kit II MW
54	Jazz Kit II
55	Hoppy Kit!
56	L'il Nipper Kit
57	Geo-Kit MW+22
58	Lo-Fi Vinyl Kit
59	Technoo Kit
60	VAST Sliders 808
61	Industry Set II
62	General MIDI Kit
63	Slam'nDrumsII GM
64	ElectroDrumsetGM
65	SmallKit+Perc MW
66	Steel Str Guitar
67	12-Str Guitar
68	Spark Guitar
69	Blue Moods
70	SloChrsGtr^Harms
71	ES335^Abercrmbie
72	SliderDistJazzGt
73	Crunchy Lead
74	NooMutes^WahTaur
75	CeeTaur^Kotolin
76	Liquid T Lead
77	Square Lead
78	AlaZawi Take 2
79	Modulead
80	Polyreal Mini
81	ModularLead1^2^3
82	Soul Boy Lead
83	Flutey Leads
84	Hrmnica^Accrdion
85	Trumpets^BrBrass
86	Hip Brass
87	Brass De ROM
88	BigBand^Hornz
89	Z-Plane Brass
90	OrcBrs^FrnchBone
91	Brt Saxy Section
92	Brt Saxy/Split
93	Mr. Parker
94	Dynasax
95	DynTrumpet^Miles
96	Wawa Trumpet MW
97	Almost Muted MW
98	Solo Trombone
99	Jazz Band
100	W.C.Flute^Winds
101	Baroque Flute
102	Synth Calliopies
103	Incan
104	String Orchestra
105	ClassicalStrings
106	StSloStr^SilkStr
107	BigStrgs^FlngStr
108	Chamber Ensemble
109	PsuedoViolin 1^2
110	Slow Solo Cello
111	Horn&Flute w/Str
112	DynOrch^WTellOrc
113	Touchy Orchestra
114	Synth Strings
115	MelloStr^ShineOn
116	Mellotron MW
117	RavStrngs^Solina
118	Choir Strings
119	CathdrVox^8veVox
120	Mixed Choir
121	NUChoir^SpaceVox
122	Marimbae^Vibeish
123	EchoBars SliderE
124	Malleflute Split
125	Malletoo
126	Highlandistic
127	Buzz Kill
128	Harshey's
129	Spunter
130	Crumper
131	Zeek
132	Elephantiasis
133	SoftMatrix12 5th
134	TchRezoid^Hungry
135	MatchStik^NuDigi
136	Peppers Digiclub
137	Frog^LowNoteTalk
138	Oink Doink
139	Wheepy Dood
140	Funk-O-Matic
141	Pulsemonster5ths
142	OB Pad^OB Brass
143	Detooner^BigPMW
144	TeknoBallCrusher
145	VTrig SquareHead
146	Razor Saw
147	Razorip Buzzsaw
148	Pulse Pass
149	Nordic Square
150	Rezonant Snob
151	Borfin'Jumps^RPT
152	Elektro Funk
153	Acid
154	Rancid
155	Fuse
156	Squwee Monosync
157	Generic Ravelead
158	Notch Sinker
159	Grungesync Sweep
160	Rezzysaws
161	Prophet Sync
162	Glider
163	Alaska Glide MW
164	Rave Classic
165	Meditator^SloEns
166	Tranquil Picked
167	The Chaser
168	Enterprize^MTree
169	Orgazmica
170	Universal
171	Asian Digital
172	Wheel Wave Sweep
173	Arystal^InTheAir
174	Cymbal Singers
175	BriteBells^Glock
176	Crystalline^RX7
177	Wave Power X 2
178	Trinibell
179	Frozen^Pizzibell
180	Den City
181	Synth Bell 1^2
182	DynPercMix^Gator
183	Workingman Space
184	OronicoKno^Shift
185	Ethereal Dawning
186	RAVReligion^Prey
187	Voyage
188	WispSingrs^Glass
189	Luscious
190	LightMist^Padify
191	VortexRev^Launch
192	Hello^A No Way
193	Environments
194	Gremlin Groupies
195	Lost In Space
196	Lunar Wind
197	Noise Toys
198	Click
199	Default Program
750	Testify
751	Prog Rock Organ
752	Syn Rock Organ
753	Dirty Syn B
754	CleanFullDrawbar
755	Loungin'
756	Mild Grunge
757	WarmDriveBallad
758	Gospel Drawbars
759	Live Drawbars
760	Keith's Revenge
761	SynthPerc Rocker
762	LORD'S Blown Amp
763	Dist GigantorGan
764	SynOrganHologram
765	SynChurch Organ
766	SynChrch&Chorale
767	Choir & Pipes
768	Theater Flutes
769	Even Tuned Bars
770	Concert Piano 2
771	Studio Grand
772	Soft Piano
773	MonoStudioGrand
774	RandomPan Grand
775	Funky Piano
776	Piano Chase
777	Grand & Elec 1
778	Grand & Elec 2
779	Detuned Piano
780	Ballad Piano
781	Piano&SpaceyPad
782	Lush Piano Pad
783	Rotating Piano
784	MajesticPianoism
785	Sonar Piano
786	Pop Grand
787	Grandsichord

Setups

1	Tonal Raver
2	Kickin' Wave
3	Hunkster Funk
4	Off Your Rocker
5	Shimmer Hop
6	Slow Hipmeister
7	Slow Driver
8	Acidic Dance
9	JR's Dance
10	Drum & Wave
11	Jetzen's Trance
12	Euro-Rave
13	Acoustic Country
14	Trip-Hopper
15	Latinease
16	Hop II
17	Slider Grooves
18	Wah Blues
19	DrumsE-B1Fills#3
20	Controlled FX
21	Quantum Mechanix
22	Driven
23	Drewz Groove
24	Jazz Session 101
25	Piano Trio & Str
26	Jazzy's Band
27	SmoothEPiano
28	Orchescape
29	New Rock Band
30	Harpsichord&Vox
31	Analogued
32	Elegant Grandeur
33	Cathedral
34	Church Organ
35	Jazz & Pop Brass
36	Glory Orchestra
37	Super Lush
38	Scoring Stage
39	Happnin' Brass
40	RbnSpltB3+MIDIPd
41	SpltVASTB+MIDIPd
42	Guitars Sent Her
43	GB 6Keys&2Basses
44	Comper Keys
45	Bold Beauty
46	Nubian Nightfall
47	Polar Reverie
48	Vapor Drawing
49	Siamese Ribbon
50	Spacegroove

51	F1 Latino Groove
52	Dream Pad RbnVel
53	C2 Jam
54	Nu C2 & Lead
55	Percolator
56	Acid Prog C2
57	Krazed Keith
58	Conn Trolled
59	Ravin & Droolin
60	Clanger
61	Waverado Rbn SW
62	Elephantitus
63	Ether Dreams
64	Rbn Slider Bells
65	Munchkin Factory
66	Clink Split
67	Lush World
68	IX Dancers
69	MidEasternBells
70	FM Morpher
71	Slo n' Funky
72	Liquid Guitars
73	Big & Meaty
74	Glazing
75	Play With Balls
76	Pepper Digital
77	De Tant
78	Old & New Pad
79	Punch-n-Oinky
80	Brazzine
81	Pad W/ Rotor
82	Tang Nebulae
83	Boink Head
84	Voyager
85	Changling
86	Wash Cycle
87	NewTaleSpinnin'
88	Electric Swirl
89	Land Of Giants
90	Desert Planet
91	Terraformation
92	Sparkle & Bass
93	Area 51 1/2
94	Slider Player
95	Synthozilla
96	Dr.Distorted
97	ControlSetup
98	Clear Setup
99	Default Setup

QA Banks

1	Keys
2	Analog
3	Hard Edge
4	Jazz
5	Rock/Blues
6	Classical
7	Film Score
8	Jukebox
9	Basic QA Bank

Songs

2	Slow Drive Arr
3	Slow Drive Sng
4	"2WaveSeqCh5,6Arr"
5	"2WaveSeqCh5,6Sng"
6	HybridGroove Arr
7	Wave Seq Ch6 Arr
8	Wave Seq Ch6 Sng
9	Funk Groove1 Arr
10	Funk Groove1 Sng
11	Funk Groove2 Arr
12	Funk Groove2 Sng
13	Funk Groove3 Sng
14	DiscoGroove1 Sng
15	DiscoGroove2 Sng
16	Funk Fill 1 Sng
17	Funk Fill 2 Sng
18	Hybrid Funk Arr
19	Hybrid Funk Sng
20	Slow Hip-Hop Arr
21	Slow Hip-Hop Sng
22	Acid Rock Arr
23	AcidRock Sng ptA
24	AcidRock Sng ptB
25	AcidRock Sng ptC
26	AcidRockSngDrums
27	Rock Groove Arr
28	Rock Groove Sng
29	UpBluesGrooveArr
30	UpBluesGrooveSng
31	AcoustiGrooveArr
32	AcoustiGrooveSng
33	Rave Groove Arr
34	Rave Groove Sng
35	Hip Groove Arr
36	Hip Groove Sng
37	Mod Groove 1 Arr
38	Mod Groove 1 Sng
39	ElectroGrooveArr
40	ElectroGrooveSng
41	Dance Groove Arr
42	Dance Groove Sng
43	WayOutGroove Arr
44	WayOutGroove Sng
45	Tek Latin Arr
46	Tek Latin Sng
47	Sound FX Arr
48	Sound FX Sng
49	Jazzmen Arr
50	Jazzmen Sng

Studios

1	RoomChorDelay Hall
2	RmChorChRv Hall
3	RoomChorCDR Hall
4	RoomChor Hall
5	RoomChrCh4T Hall
6	RoomFlngCDR Hall
7	RoomFlgEcho Hall
8	RmFlngStImg Garg
9	RmFlgChDelay Room
10	ChmbFlgGtRv Hall
11	RoomFlngCDR Hall
12	RoomFlngLsr Echo
13	RmFlgFXFlng Flng
14	SpaceFlng Hall
15	ChmbFlngCDR Verb
16	RoomPhsrCDR Hall
17	RmPhsrQuFlg Hall
18	RoomPhsr Space
19	RmEQmphEcho Comp
20	RmEQmphEcho Hall
21	RmEQmph4Tp Room
22	RmEQmph4Tap Hall
23	RmSweepEcho Hall
24	RoomResEcho Hall
25	RmRotoFl4T CmpRv
26	RoomSRSCDR Hall
27	RoomSRSRoom Room
28	RoomSRSChDl Hall
29	RoomSRSCDR CDR
30	RmStImgChDl Hall
31	RoomSRSRoom Chmb
32	RoomSRSRoom Hall
33	ChmbCompCDR Hall
34	RoomCmpChor Hall
35	RoomComp Hall
36	RoomComp Hall
37	BthComp SRS Hall
38	RoomCmpCh4T Hall
39	RmDsRotFl4t RvCm
40	RoomRmHall Hall
41	Room Room SRS2
42	RoomRmHall Hall
43	Room Room Hall
44	Room Hall Hall
45	Room Room Hall2
46	Room Room Hall3
47	Room Room Hall4
48	Room Hall Hall2
49	Sndbrd Room Hall
50	Sndbrd Rm Hall2
51	Room Room Hall3
52	AuxChrMDelay Room
53	AuxFlngChRv Room
54	AuxShp4MDelay Hall
55	AuxDistLasr Room
56	AuxEnhSp4T Class
57	AuxDistLasr Acid
58	EnhcManPhs Room
59	EnhcFlg8Tap Room
60	EnhcCmpFlng Room
61	CompEQmphCh Room
62	BthQFlg4Tap Hall
63	ChmbTremCDR Room
64	ChmbCmpFlRv Hall
65	ChamDstEcho Room
66	ChamFlg4Tap Hall
67	ChmbEnv4Tap GtRv
68	CmbrShapLsr Hall
69	AuxPtchDst+ Chmb
70	AuxChorFlRv Cmbr
71	AuxChorFlRv Cmb2
72	AuxChorFlRv Cmb3
73	AuxChorFlRv Cmb4
74	HallFlgChDl Room
75	HallPtchLsr Hall
76	HallGateFl4T Bth
77	HallChorFDR Room
78	HallPtchPtFl Lsr
79	HallFlng8Tp Room
80	HallChrEcho Room
81	HallChorCDR Hall
82	HallRsFltChDl Rm
83	Hall ChDelay Hall
84	HallFlgChDl Hall
85	Hall Room SRS
86	Hall Room Room
87	Hall CmpRvb
88	HallFlng Hall
89	HallRoomChr Hall
90	AuxPhsrFDR Hall
91	AuxChrDist+ Hall
92	AuxFlgDist+ Hall
93	AuxChrDist+ Hall
94	AuxChorMDelay Hall
95	AuxChorSp6T Hall
96	AuxChorChDl Hall
97	AuxPhasStIm Hall
98	AuxFlngCDR Hall
99	AuxPhsFlDbl Hall
100	AuxSRSRoom Hall
101	AuxFlngLasr Hall
102	AuxEnh4Tap Hall
103	EnhcChorCDR Hall
104	EnhChorChDl Hall
105	EnhcChor Plate
106	CompFlgChor Hall
107	ChorChorFlg Hall
108	ChapelSRS Hall
109	ChapelSRS Hall2
110	Chapel Room Hall
111	PltEnvFl4T Room
112	PlatEnvFl4T Filt
113	PltEnvFl4T Plate
114	PltTEnvFlg Plate
115	PlateRngMd Hall
116	AuxDist+Echo Plt
117	AuxEnvSp4T Plate
118	AuxShap4MD Plate
119	AuxChorDist+ Plt
120	AuxShFlgChDl Plt
121	AuxMPFlgLasr Plt
122	AuxShap4MD Plate
123	FlgEnv4Tap Plate
124	EnhrFlgCDR Plate
125	AuxRingPFD Plate
126	GtRvShapMDl Room
127	GtdEnhcStIm Room
128	Gtd2ChrEcho 2Vrb
129	GtdEnhcStIm Hall
130	AuxEnvSp4T GtVrb
131	GtRbSwpFlt Lasr
132	GtRbSwpFlt FlDelay
133	ChRvStIEcho Hall
134	ChorChorCDR Spac
135	ChDlDstEQ Hall
136	AuxDPanCDR ChPlt
137	AuxChorFlng CDR
138	AuxEnhcSp4T CDR
139	AuxPtchDst+ ChRv
140	EnhcChorChDl PCD
141	AuxPoly FDR
142	EnhcChorChDl FDR
143	EnhcChrChDl FDR2
144	AuxRotoSp4T FlRv
145	AuxRotaryFDR Plt
146	RotoOrgFX Hall
147	CmpRvbFlDl Hall
148	AuxEnhSp4T RvCm
149	AuxPtchRoom RvCm
150	PhsrChorCDR Phsr
151	ChDlSp4TFlDl Phs
152	AuxFlgDst+ ChLsD
153	AuxFlgDst+ ChLs2
154	RoomRoomSRS CmRv
155	RoomRoom Room
156	GtRvPlate Hall
157	RoomRoom SRS
158	EnhcSp4T Hall
159	Room RoomChr SRS
160	KB3 V/C ->Rotary
161	EQStImg 5BndEQ
162	Aux5BeqStIm Hall
163	10Band StIm Hall
164	3BndCmp PtFl
165	AuxDst+Lsr Plt
166	AuxRsFltEnvSwFlt
167	AuxChorEnvF Hall
168	AuxChRvEncr Chor
169	AuxChrDstEQ Room
170	Aux ChDlSRS Hall
171	Aux EQFlng DstEQ
172	AuxFlngPhsr Lasr
173	AuxFlShQFlg Hall
174	AuxFlgDist+ Room
175	Aux GtVbFl4T Bth
176	AuxRvRvQFlg Hall
177	AuxRvRbShapeChmb
178	AuxSpinMDelay Room
179	Aux SweepEchoBth
180	AuxRoto&DsFDRPlt

181	AuxRot&Ds2FDRPlt
182	AuxFlgChDl Hall
183	AuxDstLsr CDR
184	CPDlEnFltCmpGtRv
185	RotoOrgFX2 Hall
186	ChDlFlPtLzVb Plt
187	CDFlDelayPhRm Hall
188	CDR FlgRvb Hall
189	DstPhsPnLzVb CDR
190	DistRoom GrphEQ
191	Enh Ch 4T Hall
192	FiltCmpExpFl CDR
193	LzVbFlDstEQ Room
194	PhseDist Room
195	ChDelayRvFlRv Hall
196	RmRotr&DstChrPlt
197	Clear Studio
198	Pre-KDFX Studio
199	Default Studio
800	Sweet Hall
801	Small Hall
802	Medium Hall
803	Large Hall
804	Big Gym
805	Bright Plate 1
806	Opera House
807	Live Chamber
808	Bathroom
809	Med Large Room
810	Real Room
811	Drum Room
812	Small Dark Room
813	Small Closet
814	Add Ambience
815	Gated Reverb
816	Reverse Reverb
817	Non-Linear
818	Slapverb
819	Full Bass
820	Room + Delay
821	Delay Big Hall
822	Chorus Room
823	Chorus Smallhall
824	Chorus Med Hall
825	Chorus Big Hall
826	Chor-Delay Room
827	Chor-Delay Hall
828	Flange-Delay Room
829	Flange-Delay Hall
830	Stereo Chorus
831	Stereo Flanger
832	Stereo Delay
833	4-Tap Delay
834	Chorus Delay
835	Flange Delay
836	Chorus 4-Tap
837	Flange 4 Tap
838	Chorus Echo
839	Chorus Echoverb
840	Fast Flange
841	Wash
842	Into The Abyss
843	Space Flanger
844	Flange Room
845	Predelay Hall
846	Flange Echo
847	Rotary Club
848	Rotary Hall
849	Chorus
850	Soundbrd/rvb
851	Percussive Room
852	Brt Empty Room
853	Mosque Room
854	New Gated
855	Chorus Slap Room
856	Chorus Bass Room
857	New Chorus Hall
858	Spacious
859	Wash Lead
860	New Hall Wet/Dryelay
861	Rich Delay
862	Glass Delay
863	Real Plate
864	Real Niceverb
865	ClassicalChamber
866	Empty Stage
867	Long & Narrow
868	Far Bloom
869	Floyd Hall
870	With A Mic

Keymaps

1	Grand Piano
2	Dual Elec Piano
3	Hard Elec Piano
4	Soft Elec Piano
5	Voices
6	Ensemble Strings
7	Elec Jazz Guitar
8	Acoustic Guitar
9	5 String Guitar
10	Dual E Bass
11	Elec Pick Bass
12	Elec Slap Bass
13	Finger Atk Bass
14	Flute
15	Tenor Saxophone
16	Sax no Altissimo
17	Trumpet
18	Trombone
19	Trombone/Trumpet
20	Bone/Trp 2
21	Trombet
22	Trumpbone
23	Preview Drums
24	Dry Kit 1
25	Dry Kit 2
26	Amb Kit 1
27	Amb Kit 2
28	Amb Kit 3
29	2 8ve Dry Kit
30	General MIDI Kit
31	2-vel [1____2__]
32	3-vel [1__2__3_]
33	4-vel [12_3_4__]
34	4-vel [1_2_3_4_]
35	5-vel [1_2_3_45]
36	6-vel [1_2_3456]
37	7-vel [1_234567]
38	8-vel [12345678]
39	Ride Rim Cymbal
40	Ride Bell Cymbal
41	Crash Cymbal
42	Closed Hihat
43	Slt Open Hihat
44	Open Hihat
45	Open>Close Hihat
46	Foot Close Hihat
47	Dry Kick 1
48	Dry Kick 2
49	Amb Kick 1
50	Amb Kick 2
51	Amb Kick 3
52	DrySnare1 soft
53	DrySnare1 Hard
54	Dual Dry Snare 1
55	Dry Snare 2
56	Dry Snare 3
57	Amb Snare 1
58	Amb Snare 2
59	Amb Snare 3
60	Cross Stick

62	10in Dry Tom
63	12in Dry Tom
64	15in Dry Tom
65	13in Amb Tom
66	15in Amb Tom
67	16in Amb Tom
68	Reversals
69	Reverse Bell
71	Bidir Amb Kick 1
72	Reverse Snare
73	Conga Bass
74	Conga Slap
75	Conga Tone
76	Syn Conga Tap
77	Timbale
78	Timbale Shell
79	Cabasa
80	Clave
81	Cowbell
82	Tambourine
83	Handclaps
84	Reverse Crash
85	Reverse Clsd Hat
86	Reverse Open Hat
87	Reverse hat loop
88	Chiff
89	Chirp
90	FM Bell Trans
91	Waterphone
92	Metal Clank
93	TimbaleShell Atk
94	Cowbell Atk
95	Timbale Atk
96	Bell Attack
97	Clave Atk
98	Wood Bar Atk
99	Conga Tone Atk
100	Conga Slap Atk
101	Elec Pno Atk
102	Brass Attack
103	Bow Attack
104	Jazz Guitar Atk
105	Steel Guitar Atk
106	Perc Atk 1
107	Perc Atk 2
108	Perc Atk 3
109	Wood Bars
110	Solo Strings
111	Six String Mutes
112	Oboe Wave
113	Clav Wave
114	Elec Piano Wave
115	Bell Wave
116	Ping Wave
117	Drawbars 1-3
118	Drawbars 1-4
119	Drawbars1-3 Dist
120	Full Drawbars

121	Drawbars 1-4, 8
122	Organ Wave 1
123	Organ Wave 2
124	Organ Wave 3
125	Organ Wave 4
126	Organ Wave 5
127	Organ Wave 6
128	Organ Wave 7
129	Partials 1-3
130	Partials 4-7
131	Partials 8-12
132	Partials 13-20
133	Partials 21-30
134	Partials 1&2
135	Partials 3&4
136	Partials 5-7
137	Partials 8-10
138	Partials 11-15
139	Partials 16-21
140	Partials 2-4
141	"Partials5,7,9,11"
142	"Partials 1,2,4"
143	"Partials 1,2,4,6"
144	Partials 3-5
145	Partials 1&3
146	"Partials 1,3,5"
147	Partials 1&4
148	Partials 1&6
149	Partials 1&8
150	Partials 1&12
151	Sawtooth
152	Dull Sawtooth
153	Very Dull Saw
154	Square Wave
155	Dull Square
156	Very Dull Square
157	Buzzy Square
158	Buzz Wave
159	Hi Formant Wave
160	PrimeNumber Wave
161	Triangle Wave
162	Third Wave
163	Sine Wave
164	ExtDynPrtls1
165	ExtDynPrtls2
166	ExtDynSaw
167	Mellow Vox
168	Silence
169	Synflute Brt
170	Synflute mello
171	SlapBass/Guitar
172	Mello Vox 2
173	Shift Guitar 2
174	Single Mute
175	synElecJazzGtr
176	Sine Wave Click
177	Fingered Bass 2
178	Ext Dual Bass
179	Syn Bass Pick
180	Syn Bass Slap

181	Shift Guitar
182	Syn Guitar
183	Syn Voices
184	Syn Voices 2
185	Perc Voice
186	Synstrings 1
187	Synstrings 2
188	Syn Piano
189	Funny Perc
190	TechnoLoops
191	Hat Loop
199	Silence
770	Stereo Piano
771	Piano Left
772	Piano Right
773	Pno440 Left
774	Pno440 Right
775	Mono Piano
776	Mono Piano 440
777	Hybrid Piano 1
778	Way Dull Piano

Samples

1	Grand Piano
2	Rhodes E Piano
5	Voices
6	Ensemble Strings
7	Elec Jazz Guitar
8	Acoustic Guitar
10	Electric Bass
14	Flute
15	Tenor Saxophone
17	Trumpet/Trombone
39	Ride Rim Cymbal
40	Ride Bell Cymbal
41	Crash Cymbal
42	Closed Hihat
43	Slt Open Hihat
44	Open Hihat
45	Open>Close Hihat
46	Foot Close Hihat
47	Dry Kick 1
48	Dry Kick 2
49	Amb Kick 1
50	Amb Kick 2
51	Amb Kick 3
52	DrySnare1Soft
53	DrySnare1Hard
55	Dry Snare 2
56	Dry Snare 3
57	Ambient Snare 1
58	Ambient Snare 2
59	Ambient Snare 3
60	Cross Stick
62	10in Dry Tom
63	12in Dry Tom
64	15in Dry Tom
65	13in Amb Tom
66	15in Amb Tom
67	16in Amb Tom
68	Revrs Ride Rim
69	Revrse Ride Bell
71	Bidir Amb Kick 1
72	Revrs Amb Snre 3
73	Conga Bass
74	Conga Slap
75	Conga Tone
77	Timbale
78	Timbale Shell
79	Cabasa
80	Clave
81	Cowbell
82	Tambourine
83	Handclaps
84	Revrs Crash
85	Rvs Closed Hihat
86	Revrs Open Hihat
87	Rvrs Op Hat loop
88	Chiff
89	Chirp
90	FM Bell Trans

91	Bellhallah
92	Metal Clank
93	TimbaleShell Atk
94	Cowbell Atk
95	Timbale Atk
96	Bell Attack
97	Clave Atk
98	Wood Bar Atk
99	Conga Tone Atk
100	Conga Slap Atk
101	Elec Pno Atk
102	Brass Attack
103	Bow Attack
104	Jazz Guitar Atk
105	Steel Guitar Atk
106	Perc Atk
109	Wood Bars
110	Solo Strings
111	Six String Mutes
112	Oboe Wave
113	Clav Wave
114	Elec Piano Wave
115	Bell Wave
116	Ping Wave
117	Drawbars 1-3
118	Drawbars 1-4
119	Drawbars1-3 Dist
120	Full Drawbars
121	"Drawbars 1-4,8"
122	Organ Wave 1
123	Organ Wave 2
124	Organ Wave 3
125	Organ Wave 4
126	Organ Wave 5
127	Organ Wave 6
128	Organ Wave 7
129	Partials 1-3
130	Partials 4-7
131	Partials 8-12
132	Partials 13-20
133	Partials 21-30
134	Partials 1&2
135	Partials 3&4
136	Partials 5-7
137	Partials 8-10
138	Partials 11-15
139	Partials 16-21
140	Partials 2-4
141	"Partials5,7,9,11"
142	"Partials 1,2,4"
143	"Partials 1,2,4,6"
144	Partials 3-5
145	Partials 1&3
146	"Partials 1,3,5"
147	Partials 1&4
148	Partials 1&6
149	Partials 1&8
150	Partials 1&12

151	Sawtooth
152	Dull Sawtooth
153	Very Dull Saw
154	Square Wave
155	Dull Square
156	Very Dull Square
157	Buzzy Square
158	Buzz Wave
159	Hi Formant Wave
160	PrimeNumber Wave
161	Triangle Wave
162	Third Wave
163	Sine Wave
164	ExtDynPrtls1
165	ExtDynPart2
166	ExtDynSaw
168	Silence
174	Single Mute
175	Fingered Bass
176	Syn Bass Pick
177	Syn Bass Slap
178	Syn Guitar
179	Perc Voice
180	Syn Piano
181	Kick/Snare Loop
182	Dry Kick Loop
183	Amb Kick Loop
184	Clave/Shakr Loop
185	Xstick Loop
186	Handclap Loop
187	Hat Loop
199	Silence
770	StereoPiano b0
771	StereoPiano e1
772	StereoPiano a1
773	StereoPiano d2
774	StereoPiano a#2
775	StereoPiano d3
776	StereoPiano a3
777	StereoPiano c#4
778	StereoPiano f4
779	StereoPiano b4
780	StereoPiano f5
781	StereoPiano b5
782	StereoPiano e6
783	StereoPiano a6
784	StereoPiano a6NR
785	StereoPiano e7
786	Piano Left
787	Piano Right

FX Presets

1	NiceLittleBooth
2	Small Wood Booth
3	Natural Room
4	PrettySmallPlace
5	Sun Room
6	Soundboard
7	Add More Air
8	Standard Booth
9	A Distance Away
10	Live Place
11	Viewing Booth
15	BrightSmallRoom
16	Bassy Room
17	Percussive Room
18	SmallStudioRoom
19	ClassRoom
20	Utility Room
21	Thick Room
22	The Real Room
23	Sizzly Drum Room
24	Real Big Room
25	The Comfy Club
26	Spitty Drum Room
27	Stall One
28	Green Room
29	Tabla Room
30	Large Room
31	Platey Room
40	SmallDrumChamber
41	Brass Chamber
42	Sax Chamber
43	Plebe Chamber
44	In The Studio
45	My Garage
46	School Stairwell
47	JudgeJudyChamber
48	Bloom Chamber
55	Grandiose Hall
56	Elegant Hall
57	Bright Hall
58	Ballroom
59	Spacious Hall
60	Classic Chapel
61	Semisweet Hall
62	Pipes Hall
63	Reflective Hall
64	Smoooth Hall
65	Splendid Palace
66	Pad Space
67	Bob'sDiffuseHall
68	Abbey Piano Hall
69	Short Hall
70	The Long Haul
71	Predelay Hall
72	Sweeter Hall
73	The Piano Hall
74	Bloom Hall
75	Recital Hall
76	Generic Hall
77	Burst Space
78	Real Dense Hall
79	Concert Hall
80	Standing Ovation
81	Flinty Hall
82	HighSchool Gym
83	My Dreamy 481!!
84	Deep Hall
85	Immense Mosque
86	Dreamverb
87	Huge Batcave
95	Classic Plate
96	Weighty Platey
97	Medm Warm Plate
98	Bloom Plate
99	Clean Plate
100	Plate Mail
101	RealSmoothPlate
102	Huge Tight Plate
103	BigPredelayPlate
110	L:SmlRm R:LrgRm
111	L:SmlRm R:Hall
112	Gated Reverb
113	Gate Plate
114	Exponent Booth
115	Drum Latch1
116	Drum Latch2
117	Diffuse Gate
118	Acid Trip Room
119	Furbelows
120	Festoons
121	Reverse Reverb
122	Reverse Reverb 2
130	Guitar Echo
131	Stereo Echoes1
132	Stereo Echoes2
133	4-Tap Delay
134	OffbeatFlamDelay
135	8-Tap Delay
136	Spectral 4-Tap
137	Astral Taps
138	SpectraShapeTaps
150	Basic Chorus
151	Chorus Comeback
152	Chorusier
153	Ordinary Chorus
154	SlowSpinChorus
155	Chorus Morris
156	Everyday Chorus
157	Thick Chorus
158	Soft Chorus
159	Rock Chorus
160	Sm Stereo Chorus
161	Lg Stereo Chorus
170	Big Slow Flange
171	Wetlip Flange
172	Sweet Flange
173	Throaty Flange
174	Delirium Tremens
175	Flanger Double
176	Squeeze Flange
177	Simply Flange
178	Analog Flanger
190	Circles
191	Slow Deep Phaser
192	Manual Phaser
193	Vibrato Phaser
194	ThunderPhaser
195	Saucepan Phaser
196	Static Phaser
199	No Effect
700	Chorus Delay
701	Chorus PanDelay
702	Doubler & Echo
703	Chorus VryLngDelay
704	FastChorusDouble
705	BasicChorusDelay
706	MultiTap Chorus
707	ThickChorus no4T
708	Chorused Taps
709	Chorus Slapbacks
710	MultiEchoChorus
711	ChorusDelayHall
712	ChorDelayRvb Lead
713	ChorDelayRvb Lead2
714	Fluid ChorDelayRvb
715	ChorLite DelayHall
716	ChorusSmallRoom
717	DeepChorDelayHall
718	Chorus PercHall
719	Chorus Booth
720	ClassicEP ChorRm
721	ChorusMedChamber
722	Vanilla ChorRvb
723	Chorus Slow Hall
724	SoftChorus Hall
725	ChorBigBrtPlate
726	Chorus Air
727	Chorus HiCeiling
728	Chorus MiniHall
729	CathedralChorus
730	PsiloChorusHall
731	GuitarChorLsrDelay
732	Flange + Delay
733	ThroatyFlangeDelay
734	Flange + 4Tap
735	Bap ba-da-dap
736	Slapback Flange
737	Quantize+Flange
738	FlangeDelayHall
739	FlangeDelayRoom
740	SloFlangeDelayRoom
741	FlangeDelayBigHall
742	Flange Theatre
743	FlangeVerb Clav
744	FlangeVerb Gtr
745	Flange Hall
746	Flange Booth
747	Flange->LaserDelay
748	FlangeTap Synth
749	Lazertag Flange
750	Flange->Pitcher
751	Flange->Shaper
752	Shaper->Flange
753	Warped Echoes
754	L:Flange R:Delay
755	StereoFlamDelay
756	2Delays Ch Fl Mono
757	LaserDelay->Rvb
758	Shaper->Reverb
759	MnPitcher+Chorus
760	MnPitcher+Flange
761	Pitcher+Chor+Delay
762	Pitcher+Flng+Delay
763	SubtleDistortion
764	Synth Distortion
765	Dist Cab EPiano
766	Distortion+EQ
767	Burnt Transistor
768	TubeAmp DelayChor
769	TubeAmp DelayChor2
770	TubeAmp DelayFlnge

771	TubeAmp Flange
772	PolyAmp Chorus
773	PolyAmp DelayFlnge
774	VibrChor Rotors
775	SlightDistRotors
776	Rotostort
777	VibrChor Rotors2
778	Full VbCh Rotors
779	KB3 FXBus
780	KB3 AuxFX
781	Pitch Spinner
782	VibrChrDstRotor1
783	VibrChrDstRotor2
784	VibChrDstRotor3
785	FullVbChTubeRotr
786	ChorDelayHall 2
787	Flange Hall 2
788	SpeeChorusDeep
789	Fluid Wash
790	VC+DistRotor
800	Sweet Hall
801	Small Hall
802	Medium Hall
803	Large Hall
804	Big Gym
805	Bright Plate 1
806	Opera House
807	Live Chamber
808	Bathroom
809	Med Large Room
810	Real Room
811	Drum Room
812	Small Dark Room
813	Small Closet
814	Add Ambience
815	Gated Reverb
816	Reverse Reverb
817	Non-Linear
818	Slapverb
819	Full Bass
820	Room + Delay
821	Delay Big Hall
822	Chorus Room
823	Chorus Smallhall
824	Chorus Med Hall
825	Chorus Big Hall
826	Chor-Delay Room
827	Chor-Delay Hall
828	Flange-Delay Room
829	Flange-Delay Hall
830	Stereo Chorus
831	Stereo Flanger
832	Stereo Delay
833	4-Tap Delay
834	Chorus Delay
835	Flange Delay
836	Chorus 4-Tap
837	Flange 4 Tap
838	Chorus Echo
839	Chorus Echoverb
840	Fast Flange
841	Wash
842	Into The Abyss
843	Space Flanger
844	Flange Room
845	Predelay Hall
846	Flange Echo
847	Rotary Club
848	Rotary Hall
849	Chorus
850	Soundbrd/rvb
851	Percussive Room
852	Brt Empty Room
853	Mosque Room
854	New Gated
855	Chorus Slap Room
856	Chorus Bass Room
857	New Chorus Hall
858	Spacious
859	Wash Lead
860	New Hall Wet/Dryelay
861	Rich Delay
862	Glass Delay
863	Real Plate
864	Real Niceverb
865	ClassicalChamber
866	Empty Stage
867	Long & Narrow
868	Far Bloom
869	Floyd Hall
870	With A Mic
900	Basic Env Filter
901	Phunk Env Filter
902	Synth Env Filter
903	Bass Env Filter
904	EPno Env Filter
905	Trig Env Filter
906	LFO Sweep Filter
907	DoubleRiseFilter
908	Circle Bandsweep
909	Resonant Filter
910	Dual Res Filter
911	EQ Morpher
912	Mono EQ Morpher
913	Ring Modulator
914	PitcherA
915	PitcherB
916	SuperShaper
917	SubtleDrumShape
918	3 Band Shaper
919	LaserVerb
920	Laserwaves
921	Crystallizer
922	Spry Young Boy
923	Cheap LaserVerb
924	Drum Neurezonate
925	LazerfazerEchoes
926	Simple Lazerverb
927	TripFilter
950	HKCompressor 3:1
951	DrumKompress 5:1
952	SK FB Comprs 6:1
953	SKCompressor 9:1
954	SKCompressr 12:1
955	Compress w/SC EQ
956	Compress/Expand
957	Comprs/Expnd +EQ
958	Reverb>Compress
959	Reverb>Compress2
960	Drum Comprs>Rvb
961	Expander
962	3Band Compressor
963	Simple Gate
964	Gate w/ SC EQ
965	Graphic EQ
966	5 Band EQ
967	ContourGraphicEQ
968	Dance GraphicEQ
969	OldPianoEnhancer
970	3 Band Enhancer
971	3 Band Enhancer2
972	Extreem Enhancer
973	Tremolo
974	Dual Panner
975	SRS
976	Widespread
977	Mono->Stereo
978	3 Band Compress
979	Simple Panner
980	Big Bass EQ
998	Stereo Analyze
999	FX Mod Diag

FX Algorithms

1	MiniVerb
2	Dual MiniVerb
3	Gated MiniVerb
4	Classic Place
5	Classic Verb
6	TQ Place
7	TQ Verb
8	Diffuse Place
9	Diffuse Verb
10	OmniPlace
11	OmniVerb
12	Panaural Room
13	Stereo Hall
14	Grand Plate
15	Finite Verb
130	Complex Echo
131	4-Tap Delay
132	4-Tap Delay BPM
133	8-Tap Delay
134	8-Tap Delay BPM
135	Spectral 4-Tap
136	Spectral 6-Tap
150	Chorus 1
151	Chorus 2
152	Dual Chorus 1
153	Dual Chorus 2
154	Flanger 1
155	Flanger 2
156	LFO Phaser
157	LFO Phaser Twin
158	Manual Phaser
159	Vibrato Phaser
160	SingleLFO Phaser
700	Chorus+Delay
701	Chorus+4Tap
702	Chorus<>4Tap
703	Chor+Delay+Reverb
704	Chorus<>Reverb
705	Chorus<>LasrDelay
706	Flange+Delay
707	Flange+4Tap
708	Flange<>4Tap
709	Flan+Delay+Reverb
710	Flange<>Reverb
711	Flange<>LasrDelay
712	Flange<>Pitcher
713	Flange<>Shaper
714	Quantize+Flange
715	Dual MovDelay
716	Quad MovDelay
717	LasrDelay<>Reverb
718	Shaper<>Reverb
719	Reverb<>Compress
720	MonoPitcher+Chor

721	MonoPitcher+Flan
722	Pitcher+Chor+Delay
723	Pitcher+Flan+Delay
724	Mono Distortion
725	MonoDistort+Cab
726	MonoDistort + EQ
727	PolyDistort + EQ
728	StereoDistort+EQ
729	TubeAmp<>MD>Chor
730	TubeAmp<>MD>Flan
731	PolyAmp<>MD>Chor
732	PolyAmp<>MD>Flan
733	VibChor+Rotor 2
734	Distort + Rotary
735	KB3 FXBus
736	KB3 AuxFX
737	VibChor+Rotor 4
738	VC+Dist+1Rotor 2
739	VC+Dist+HiLoRotr
740	VC+Tube+Rotor 4
900	Env Follow Filt
901	TrigEnvelopeFilt
902	LFO Sweep Filter
903	Resonant Filter
904	Dual Res Filter
905	EQ Morpher
906	Mono EQ Morpher
907	Ring Modulator
908	Pitcher
909	Super Shaper
910	3 Band Shaper
911	Mono LaserVerb
912	LaserVerb Lite
913	LaserVerb
950	HardKneeCompress
951	SoftKneeCompress
952	Expander
953	Compress w/SC EQ
954	Compress/Expand
955	Comp/Exp + EQ
956	Compress 3 Band
957	Gate
958	Gate w/ SC EQ
959	2 Band Enhancer
960	3 Band Enhancer
961	Tremolo
962	Tremolo BPM
963	AutoPanner
964	Dual AutoPanner
965	SRS
966	Stereo Image
967	Mono -> Stereo
968	Graphic EQ
969	Dual Graphic EQ
970	5 Band EQ
998	FXMod Diagnostic
999	Stereo Analyze

Program Control Assignments

ID	Name		
1	Concert Piano 1	MIDI 25	(Aux) Hall Level+Time
		MIDI 29	Soundboard Wet/Dry
			Soft Pedal is active
2	Stereo Solo Pno	Data	InEQ: Treb
		MIDI 25	(Aux) Hall Level+Time
		MIDI 29	Soundboard Wet/Dry
			Soft Pedal is active
3	Brt Concert Pno	Data	InEQ: Treb
		MIDI 25	(Aux) Hall Level+Time
		MIDI 29	Soundboard Wet/Dry
			Soft Pedal is active
4	Rok Piano	MIDI 25	Room Wet/Dry
		MIDI 26	(Aux) Hall Level
		MIDI 27	Hall Time
5	Piano for Lyrs	MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time
		MIDI 29	Soundboard Wet/Dry
6	DrkPno^ArakisPno	MWheel	Vibrato (ArakisPno)
		Data	Toggle: DrkPno ^ ArakisPno
		MIDI 22	detune
		MIDI 25	(Aux) Chorus/Plate Level + Wet/Dry
		MIDI 26	Plate Time
		MIDI 27	Chorus FB
		MIDI 28	Chorus Mix
		MPress	Vibrato (ArakisPno)
7	Honky-Tonk	MWheel	Tremolo
		MIDI 25	(Aux) Hall Time
		MIDI 26	(Aux) Chorus Mix
		MIDI 27	Chorus FB
		MIDI 28	(Aux) Delay Mix
		MIDI 29	Delay Time adj
8	Pno&Syn/AcString	MWheel	String Balance
		Data	String Balance
		MIDI 22	Toggle: SynStrings/AcStrings
		MIDI 25	"(Aux) Hall Level, Room Wet/Dry(Layer 1 pno)"
		MIDI 26	Room Time
		MIDI 27	"EQ Treb boost, SRS Space"
		MIDI 28	"(Aux) Hall, Room HFDamp"
9	ClassicPiano&Vox	MWheel	Vox Level
		Data	"Vox Balance, Piano Treb boost"
		MIDI 22	Vox EQ Bass
		MIDI 23	"Vox EQ Treb, St Image Mix"
		MIDI 25	"(Aux) Hall Level, Room Wet/Dry"
		MIDI 26	Room and Hall Times
		MIDI 27	St Image In Gain
		MIDI 28	St Image CenterGain
		MIDI 29	Vox St Image L/R Delay
10	E Grand Stack	MWheel	String Level
		Data	InEQ: Treb boost
		MIDI 25	"(Aux) Room Level, (Aux) FDR Wet/Dry"
		MIDI 26	Flange Mix
		MIDI 27	Flange Tempo
		MIDI 28	Enhc Lo/Mid Drive
		MIDI 29	FDR Delay Mix adj

ID	Name		
11	Piano D'Biggy	MWheel	Vibrato
		Data	"BandPass adj, EnvCtl: Rel"
		MIDI 22	(Aux) Chorus Mix
		MIDI 23	(Aux) Delay Mix
		MIDI 24	Delay FB
		MIDI 25	(Aux) Hall Mix
		MIDI 26	Hall dry Mix+Time
		MIDI 27	InEQ: Treb
		MIDI 28	InEQ: Bass
		MIDI 29	Chorus Pitch Env adj
12	DynEPiano^EPnoPF	MWheel	Tremolo/ Vibrato
		Data	"Toggle: DynEPiano ^ EPnoPF, Chorus LFODepth+Rate, (Aux) Plate Level cut+PreDelay adj"
		MIDI 22	Chorus Wet/Dry
		MIDI 23	Chorus LFODepth
		MIDI 24	Chorus Xcouple
		MIDI 25	(Aux) Plate Wet/Dry+Decay Time
		MIDI 26	Plate Room Size
		MIDI 27	Chorus FB
		MIDI 28	Chorus Tap Level
		MIDI 29	Chorus Rate adj
13	FM-ish E-Piano	MWheel	"Layer detune, Vibrato, Saw+Shp detune"
		Data	Layer Enable
		MIDI 22	Layer detune
		MIDI 23	Layer octave shift
		MIDI 24	"Layer detune, Saw+Shp detune"
		MIDI 25	"CDR Wet/Dry, (Aux) Hall Wet/Dry+Time"
		MIDI 26	"Enhc Xover, Chorus Wet/Dry"
		MIDI 27	Chorus FB and Rate
		MIDI 28	CDR Chorus Depth
14	Suitcase E Piano	MWheel	Tremelo Depth
		Data	Tremelo Rate
		MIDI 25	(Aux) Room Level
		MIDI 26	Enhc Level
		MIDI 27	Flange Wet/Dry
		MIDI 28	"Enhc Hi Drive, Flange FB"
		MIDI 29	"Toggle: Enhc + Flange, EQ: Treb"
		BreathCtl	LoPass Freq and Res
15	PhunkEPno^FonkMW	MWheel	Tremolo ^ Filt and Res Depth
		Data	Toggle: PhunkEPno ^ FonkMW
		MIDI 22	Notch Filt Freq ^ Tremolo
		MIDI 23	"InEQ: Treb boost, Lowpass Depth ^ Tremelo Rate"
		MIDI 24	"Chrous Level, Notch Width ^ EnvCtl"
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time
		MIDI 27	4Tapap Wet/Dry
		MIDI 28	4Tapap FB
		MIDI 29	4Tapap I/O

ID	Name		
16	Clavinators 1^2	MWheel	detune/ LFO
		Data	"Toggle: Clavinators1 ^ 2 , EnvCtl: Att+Dec (Layer 8)"
		MIDI 22	"Layer Enable, Band/Lo/HiPass adj, Shaper amt"
		MIDI 23	EnvCtl: Att+Dec
		MIDI 24	"EnvCtl: Rel, (Aux) Hall Time"
		MIDI 25	Chorus FB
		MIDI 26	Chorus Rate+Depth
		MIDI 27	Delay Mix
		MIDI 28	Delay Time
		MIDI 29	Toggle: Room I/O
17	Spaceychord 1^2	MWheel	"Vibrato, Phaser LFO/Ctr Freq"
		Data	Toggle: Spaceychord 1 ^ 2
		MIDI 22	"SP4Tap Wet/Dry, Bass Freq+Res+boost"
		MIDI 23	"4Tap FB, BandPass Freq adj"
		MIDI 24	"4Tap Tempo, EnvCtl: Rel"
		MIDI 25	ChorDelay Wet/Dry
		MIDI 26	(Aux) Phase Notch
		MIDI 27	Phase FB
		MIDI 28	4Tap FB Image
		MIDI 29	Phase Level adj
18	DaNU Clavs 1^2^3	MWheel	Layer balance
		Data	Toggle: DaNU Clavs 1 ^ 2 ^ 3
		MIDI 22	Flange Wet/Dry
		MIDI 23	Flange LFO Level
		MIDI 24	EnvCtl: Att+Rel
		MIDI 25	(Aux) Chorus Slapback Level
		MIDI 26	Chorus Mix
		MIDI 27	"Flange Tempo, FB"
		MIDI 29	"Toggle: Dist/ Flange, Layer balance"
19	VAST B3! SW2+CC2	MWheel	Drawbar 9
		Data	Drawbar 1
		MIDI 22	Drawbar 2
		MIDI 23	Drawbar 3
		MIDI 24	"Drawbar 4, EnvCtl: Imp"
		MIDI 25	Drawbar 5
		MIDI 26	Drawbar 6
		MIDI 27	Drawbar 7
		MIDI 28	EnvCtl: Imp
		MIDI 29	Toggle: Vib/Chorus I/O
		Breath	"(Aux) Plate Level, Dist Drive+adj, EQ Bass+Treb"
		PWheel	Leslie Depth
20	Rock Organ	MWheel	Leslie Depth
		Data	Drawbar 1
		MIDI 22	Drawbar 2
		MIDI 23	"Drawbar 3, "
		MIDI 24	"Drawbar 4, (Aux) Plate Time"
		MIDI 25	KeyClick
		MIDI 26	Perc Harmonic (Hi/Low)
		MIDI 27	"HFDamp, Perc Decay"
		MIDI 28	Plate Level
		MIDI 29	Toggle: VibeChorus I/O

ID	Name		
21	Ballad Organ	MWheel	Leslie Depth
		Data	Drawbar 1
		MIDI 22	Drawbar 2
		MIDI 23	"Drawbar 3, (Aux) Plate Level"
		MIDI 24	"Drawbar 4, Plate Time"
		MIDI 25	KeyClick
		MIDI 26	Perc Harmonic (Hi/Low)
		MIDI 27	"HFDamp, Perc Decay"
		MIDI 28	Cabinet Dist Drive+Lopass
		MIDI 29	Toggle: VibeChorus I/O
22	Gospel Organ	MWheel	Leslie Depth
		Data	Drawbar 1
		MIDI 22	Drawbar 2
		MIDI 23	"Drawbar 3, (Aux) Plate Level"
		MIDI 24	"Drawbar 4, Plate Time"
		MIDI 25	KeyClick
		MIDI 26	Perc Harmonic (Hi/Low)
		MIDI 27	"HFDamp, Perc Decay"
		MIDI 28	Cabinet Dist Drive + Lopass adj
		MIDI 29	Toggle: VibeChorus I/O
23	OverDrive Organ	MWheel	Leslie Depth
		Data	Drawbar 1
		MIDI 22	Drawbar 2
		MIDI 23	"Drawbar 3, (Aux) Plate Level"
		MIDI 24	"Drawbar 4, Plate Time"
		MIDI 25	KeyClick
		MIDI 26	Perc Harmonic (Hi/Low)
		MIDI 27	"HFDamp, Perc Decay"
		MIDI 28	Cabinet Dist Drive+Lopass adj
		MIDI 29	Toggle: VibeChorus I/O
24	Perc Organ	MWheel	Leslie Depth
		Data	Drawbar 1
		MIDI 22	Drawbar 2
		MIDI 23	"Drawbar 3, (Aux) Plate Level"
		MIDI 24	"Drawbar 4, Plate Time"
		MIDI 25	KeyClick
		MIDI 26	Perc Harmonic (Hi/Low)
		MIDI 27	"HFDamp, Perc Decay"
		MIDI 28	Cabinet Dist Drive+Lopass adj
		MIDI 29	Toggle: VibeChorus I/O
25	Chiffy Pipes	MWheel	Decrescendo
		Data	LoPass Freq
		MIDI 22	Key Click
		MIDI 23	Vibrato
		MIDI 25	(Aux) Hall Level+Wet/Dry
		MIDI 26	Hall Time
		MIDI 27	Hall Early Ref Level
		MIDI 29	"Toggle: Chorus I/O, Hall HFDamp+PreDelay"
		MPress	Vibrato
26	Offertory	MWheel	Layer Balance
		Data	LoPass Freq
		MIDI 22	Layer Balance
		MIDI 23	InEQ: Treb boost
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Wet/Dry
		MIDI 27	Hall Time
		MIDI 28	Hall EarlyRefLevel
		MIDI 29	"Chorus I/O, EQ Mid"

ID	Name		
27	Pedal Pipes	MWheel	"DecRescendo, Slight Vibrato"
		Data	Layer Balance
		MIDI 25	(Aux) Hall Level+Wet/Dry
		MIDI 26	Hall Time
		MIDI 27	Hall EarlyRefLevel
		MIDI 28	Chorus FB
		MIDI 29	"Chorus I/O, Hall adj"
28	Church Organ	MWheel	Vibrato
		Data	Layer 1 Xfade
		MIDI 22	Layer Balance
		MIDI 23	Layer Balance
		MIDI 25	(Aux) Hall Wet/Dry
		MIDI 26	Hall Time
		MIDI 27	Hall EarlyRefLevel
		MIDI 29	Toggle: Hall Mix
		MPress	Vibrato
29	DualBass^SlpBass	MWheel	Vibrato
		Data	Toggle: DualBass + SlpBass
		MIDI 22	"EnvCtl: Dec, BandPass adj, ParaTreb"
		MIDI 23	EnvCtl : Att+Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Room Level+Time
		MIDI 26	Phaser Notch/ BP ^ Enhc LoDrive+Delay
		MIDI 27	Phaser Center Freq L ^ Enhc Hi Mix
		MIDI 28	Phaser Center Freq R ^ Enhc Mid Mix
		MIDI 29	Phaser FB boost * Enhc Crossover Freq
		MPress	Vibrato
30	Warm Basses	MWheel	Vibrato
		Data	Toggle: Layers
		MIDI 22	"LoPass adj, Shaper amt, EnvCtl: Imp+Att"
		MIDI 23	"EnvCtl: Imp, ParaBass+HighPass Freq"
		MIDI 24	"EnvCtl: Rel, InEQ: Bass"
		MIDI 25	(Aux) Room Level
		MIDI 26	Room Absorption
		MIDI 27	Comp Ratio
		MIDI 28	Comp: Att+Rel Time
		MIDI 29	add EQ Morph
		MPress	Vibrato
31	SustBass^MixBass	MWheel	"Vibrato, LoPass Freq"
		Data	Toggle: SustBass + MixBass
		MIDI 22	"BandPass Freq+Width, EnvCtl: Imp, LoPass adj"
		MIDI 23	EnvCtl: Rel
		MIDI 24	In EQ: Bass
		MIDI 25	Comp Att Time
		MIDI 26	Comp Rel Time
		MIDI 27	Comp Ratio
		MIDI 28	Comp ThReshhold
		MIDI 29	"Toggle: Comp I/O, (Aux) Room I/O"
		MPress	Vibrato

ID	Name		
32	RickenBass	MWheel	Vibrato
		Data	Layer Enable
		MIDI 22	Bass Cut
		MIDI 23	EnvCtl: Att+Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	"Flange Wet/Dry, Chorus Wet/ Dry"
		MIDI 27	"Flange FB+Level, Chorus FB"
		MIDI 28	"Flange L/R phase, Chorus Rate"
		MIDI 29	Toggle: Flange + Chorus
		MPress	Vibrato
33	Synth Fretless	MWheel	"Vibrato,"
		Data	"Shaper amt, HiPass Freq"
		MIDI 22	InEQ: Bass
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	"Flange Wet/Dry, Chorus Wet/ Dry"
		MIDI 27	"Flange FB, Chorus FB"
		MIDI 28	"Flange L/R Phase, Chorus Rate"
		MIDI 29	Toggle: Flange + Chorus
		MPress	"Vibrato, Shaper adj, Flange Wet/Dry"
34	Moogy Bass 1^2	MWheel	Vibrato
		Data	Toggle: Moogy Bass 1 ^ 2
		MIDI 22	"LoPass Filt adj, EnvCtl: Att"
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Place Level
		MIDI 26	"Flange Wet/Dry(sys), StImg EQ Bass"
		MIDI 27	"Flange FB, StImg CenterGain"
		MIDI 28	"Place HFDamp, Flange HFDamp, StImg L/R Delay"
		MIDI 29	Toggle: Flange + Widespread (St.Img)
		MPress	Vibrato
35	Moogy Bass 3^4	MWheel	Vibrato
		Data	Toggle: Moogy Bass 3 ^ 4
		MIDI 22	"LoPass Freq, EnvCtl: Imp"
		MIDI 23	LoPass Res
		MIDI 24	EnvCtl: Imp+Rel
		MIDI 25	"(Aux) Chorus Level+Wet/Dry, (fx2) Room Cut"
		MIDI 26	"(fx2)Chorus Mix, Enhc Crossover 1"
		MIDI 27	"Chorus FB, Enhc Crossover 2"
		MIDI 28	"Room HFDamp, Enhc Drive adj"
		MIDI 29	Toggle: ChorVerb + Enhc; Enhc Lo+Mid+Hi Drive
		MPress	Vibrato

ID	Name		
36	Matrix Big Bass	MWheel	Vibrato
		Data	Layer Toggle
		MIDI 22	"EnvCtl: Imp+Att, LoPass adj"
		MIDI 23	EnvCtl: Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Room Level+HFDamp
		MIDI 26	"Flange Wet/Dry, Tube Drv"
		MIDI 27	"Flange LFO period, MDelay Wet/Dry"
		MIDI 28	"Flange FB, MDelay FB, Dist Warmth"
		MIDI 29	Toggle: Flange + TubeAmpDelayCh
		MPress	Vibrato
37	Pitt Bass mono 1^2	MWheel	Vibrato
		Data	"Layer Enable, EnvCtl: Rel"
		MIDI 22	"PWM Filt+Width adj, EnvCtl: Att+Dec, ParaBass Frq"
		MIDI 23	"InEQ: Bass, EnvCtl: Imp"
		MIDI 24	"InEQ: Treb, EnvCtl: Rel"
		MIDI 25	(Aux) Chamber Mix
		MIDI 26	Chamber HFDamp+Time
		MIDI 27	Chorus FB
		MIDI 28	Delay Mix
		MIDI 29	Chorus Depth
		MPress	Vibrato
38	Punch Bass 1^2	MWheel	"Vibrato, LoPass Res"
		Data	Toggle: Punch Bass 1 ^ 2
		MIDI 22	InEQ: Bass
		MIDI 23	EnvCtl: Imp+Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Chamber Mix+Level+Time+HFDamp
		MIDI 26	InEQ: Treb
		MIDI 27	Chorus FB
		MIDI 28	Chorus Depth
		MIDI 29	Flange I/O
		MPress	Vibrato
39	Tee Bee This MW	MWheel	LoPass Freq
		Data	LoPass Res
		MIDI 22	EnvCtl: Imp
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level+adj
		MIDI 26	Chorus Wet/Dry
		MIDI 27	Chorus FB
		MIDI 28	Chorus Tap Pan
		MIDI 29	add Enhc
		MPress	Vibrato
40	BottomFeed^Pulse	MWheel	Vibrato
		Data	Toggle: BottomFeed ^ Pulse
		MIDI 22	"LoPass Gate+Freq, EnvCtl: Imp+Att"
		MIDI 23	"EnvCtl: Att+Dec, Saw Pitch adj"
		MIDI 24	EnvCtl: Rel
		MIDI 25	"(Aux) Room Level, (FX3)Hall Mix"
		MIDI 26	Chorus Mix
		MIDI 27	Chorus Rate
		MIDI 28	Chorus FB
		MIDI 29	Toggle: Chorus(4Tap) + Flange
		MPress	Vibrato

ID	Name		
41	SkoolBass^SImple	MWheel	Vibrato
		Data	Toggle: SkoolBass ^ SImple
		MIDI 22	"Pulse Width+Freq, Pitch adj, EnvCtl: Imp+Att"
		MIDI 23	"Dist Drive adj, EnvCtl: Dec"
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Room Level
		MIDI 26	"Phase Notch/ Dry, Dist Wet/Dry"
		MIDI 27	"Phase Center Freq, Dist Drive adj"
		MIDI 28	"Phase LFO Depth, Dist Bass adj"
		MIDI 29	"Toggle: Phase + Dist, Room Time adj"
		MPress	Vibrato
		AttVel	LoPass gate
		GKeyNum	L/R Phase
42	MonoBass^FixBass	MW	Vibrato
		Data	"Toggle: MonoBass ^ FixBass, BandPass Width"
		MIDI 22	"LoPass Freq, ^ Notch Freq+Width, EnvCtl: Imp"
		MIDI 23	"InEQ: Bass, EnvCtl: Att"
		MIDI 24	"InEQ: Treb, Hall HFDamp, EnvCtl: Rel"
		MIDI 25	(Aux) CDR Level+Hall Time
		MIDI 26	Delay Mix
		MIDI 27	Phaser FB Cut
		MIDI 28	"Phaser LFO Rate, Hall Mix"
		MIDI 29	"Chorus-Delay Cut, Phase Notch adj"
		MPress	Vibrato
43	Ravelike Bass	MWheel	Vibrato
		Data	Later Enable
		MIDI 22	"LoPass Res, EnvCtl: Att"
		MIDI 23	"InEQ: Bass, EnvCtl: Imp"
		MIDI 24	"InEQ: Treb, EnvCtl: Dec+Rel"
		MIDI 25	(Aux) Room Level
		MIDI 26	Flange Mix (sys)
		MIDI 27	"Pitcher Mix, LsrDelay HFDamp"
		MIDI 28	"Pitcher Pitch, LsrDelay Time"
		MIDI 29	Toggle: Laserverb I/O
		MPress	Vibrato
44	Junosis^Geo-Bass	MWheel	Oscilator Vibrato
		Data	"LoPass Freq, Layer Enable"
		MIDI 22	LoPass Res
		MIDI 23	"Pitch Env, EnvCtl: Rel"
		MIDI 24	EnvCtl: Imp
		MIDI 25	EnvFlt Wet/Dry
		MIDI 26	EnvFlt Min Freq
		MIDI 27	InEQ: Bass
		MIDI 28	InEQ: Treb
		MIDI 29	Toggle: (Aux) SweepFlt I/O
		MPress	Sweep Res
		PWheel	(+/ -) 2 octaves

ID	Name		
45	Ace Bass^ChirpBas	MWheel	Vibrato
		Data	"Toggle: Ace Bass ^ Chirp Bass, LoPass Filt+Res"
		MIDI 22	"HiPass Freq, LoPass Res, EnvCtl: Imp"
		MIDI 23	"LoPass Res, EnvCtl: Att"
		MIDI 24	EnvCtl: Att+Rel
		MIDI 25	(Aux) Room Level
		MIDI 26	"Flange Wet/Dry, Chorus Wet/Dry"
		MIDI 27	"Flange FB, Chorus FB"
		MIDI 28	"Flange LFO Period, Chorus Tap Delay"
		MIDI 29	Toggle: Flange + Chorus
		MPress	Vibrato
		PW	Octave Shift
46	Gritz	MWheel	Vibrato + Shaper Pitch Cut
		Data	LoPass Freq
		MIDI 22	LoPass Res
		MIDI 23	"Pitch ASR, EnvCtl: Imp"
		MIDI 24	EnvCtl: Release
		MIDI 25	"(Aux) Room Level, Laser FB"
		MIDI 26	"LsrDelay Time, TubeDrive adj"
		MIDI 27	"LsrDelay Spacing, Tube Warmth"
		MIDI 28	LsrDelay Contour
		MIDI 29	Toggle: LaserDelay + TubeDrive
		MPress	Vibrato
47	Homey Saw	MWheel	Vibrato
		Data	Pitch
		MIDI 22	"LoPass Res, Freq, Chorus Cut, EnvCtl: Imp"
		MIDI 23	"InEQ: Bass, EnvCtl: Att+Dec"
		MIDI 24	"InEQ: Treb, EnvCtl: Rel"
		MIDI 25	(Aux) Hall Level
		MIDI 26	Chorus FB
		MIDI 27	Delay Mix
		MIDI 28	Delay Time
		MIDI 29	Hall Decay Time+Room size
		MPress	Vibrato
48	OG	MWheel	Sync Amp (AClock)
		Data	Sync M Pitch
		MIDI 22	Sync S Pitch
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Plate Level
		MIDI 26	Plate Time+HFDamp
		MIDI 27	Tube Drive
		MIDI 28	MDelay Wet/Dry
		MIDI 29	TubeDelayCh Wet/Dry
49	Lowdown Bass	MWheel	"Vibrato, HiPass Freq (Chirp)"
		Data	LoPass Gate
		MIDI 22	EnvCtl: Imp
		MIDI 23	EnvCtl: Att
		MIDI 24	"Layer Enable, EnvCtl: Dec+Rel"
		MIDI 25	(Aux) Dist Level
		MIDI 26	"Dist Drive, Mid EQ cut, Flange Wet/Dry"
		MIDI 27	"InEQ: Bass, Flange FB"
		MIDI 28	Cab HiPass
		MIDI 29	Toggle: EQ + Flange
		MPress	Vibrato
50	SquashStudio Kit	MWheel	AltControl: Toms
		Data	"Pitch: Kicks, Snares, Toms, HiHats"
		MIDI 22	Snare Filters
		MIDI 23	Kick Filters
		MIDI 24	"EnvCtl: Kicks, Snares, Toms"
		MIDI 25	"(FX1+2)- (Aux) Room Level+Time, (FX2)- Mix Level "
		MIDI 26	(FX2) Compressor Ratio+Gain
		MIDI 27	Room HFDamp
		MIDI 28	"Toggle: Enhancer HiDrive, Room PreDelay"
		MIDI 29	Enhancer Hi Delay Time
51	Retro Skins MW	MWheel	Multiple Layer Toggle
		Data	Pitch: Kicks
		MIDI 22	Pitch: Snares
		MIDI 23	"Filter Freq: Kicks, Toms, Ride, AuxPerc "
		MIDI 24	EnvCtl: Kicks+Snares
		MIDI 25	(FX1+2) Rooms Wet/Dry
		MIDI 26	(Aux) Room Wet/Dry
		MIDI 27	(Aux) Compressor Attack Time
		MIDI 28	(FX1) InEQ: Bass+Treb
		MIDI 29	Toggle: Alien Skin Effect
52	2 Live Kits 2 MW	MWheel	Multiple Layer Toggle
		Data	"Pitch: Kicks, Toms"
		MIDI 22	Pitch: Snares
		MIDI 23	"HF Stimulator: Cymbal, HiHats"
		MIDI 24	"EnvCtl: Kicks, Snares, Toms, Cymbal"
		MIDI 25	"(FX1)-(Aux) Hall Level, (FX2) Plate PreDelay"
		MIDI 26	(FX2)-(Aux) Hall Level
		MIDI 27	(FX1) GateRvb Wet/Dry+Gate Threshold
		MIDI 28	"Hall Time, Plate Wet/Dry"
		MIDI 29	Toggle: Plate RvrbTime boost-Megaverb!
53	Garage Kit II MW	MWheel	Multiple Layer Toggle
		Data	"Pitch: Kicks, Toms"
		MIDI 22	"Pitch: Snares, Crash2"
		MIDI 23	"EnvCtl: Kicks, Toms"
		MIDI 24	"EnvCtl: Snares, HiHats"
		MIDI 25	(Aux) RoomGate Absorption+Gain
		MIDI 26	(FX3) Compression control
		MIDI 27	(FX3) InEQ: Treb
		MIDI 28	(FX3) InEQ: Bass
		MIDI 29	"Toggle: (Aux) Room type, Lopass adj"
54	Jazz Kit II	MWheel	Pitch: AuxPerc
		Data	"Pitch: Kicks, Toms"
		MIDI 22	Pitch: Snares
		MIDI 23	"Gain: HiHats, Crash Cym"
		MIDI 24	"EnvClt: Kicks, Toms"
		MIDI 25	(FX1+2) Rooms Wet/Dry+Time
		MIDI 26	"(FX1+2)- (Aux) Hall Level, (FX2)- Mix Level"
		MIDI 27	(FX2) In EQ: Treb cut
		MIDI 28	(Aux) Hall TrebShlf Freq+cut

ID	Name		
55	Hoppy Kit!	MWheel	"Notchy Filter: Toms, CrossStick"
		Data	"Pitch: Kicks, Toms, HiHats, Cowbell"
		MIDI 22	"Pitch: Snares, Crash1, Ride, Clap, Clave, Tambourine"
		MIDI 23	"Filter: Kicks, Snare2-3, Crash2, Ride; [Impact on Kick2]"
		+MW	"Filter: Snare1, HiHats, Crash1, AuxPerc"
		MIDI 24	"EnvCtl: Toms, Kick1, Snares"
		MIDI 25	"(FX1) Room Wet/Dry, (FX2) Flange Wet/Dry"
		MIDI 26	(Aux) Booth Level
		MIDI 27	Booth Absorption
		MIDI 28	"Room HFDamp, Flange FB"
		MIDI 29	Toggle: Room + Flange
56	L'il Nipper Kit	MWheel	SFX Pitch
		Data	"Pitch: Kick, Toms"
		MIDI 22	"Pitch: Snares, AuxPerc"
		MIDI 23	"Filter: Hihats, Cymbals"
		MIDI 24	"EnvCtl: Kicks, Snares"
		MIDI 25	(Aux) Plate Time
		MIDI 26	(FX3) Laserverb Spacing
		MIDI 27	"(FX2) Pitcher Pitch, Pitcher Wet/Dry"
		MIDI 28	Pitcher control
		MIDI 29	Laserverb Delay+Contour+FB
		MPress	AuxPerc Pitch
57	Geo-Kit MW+22	MWheel	Multiple Layer Toggle
		Data	"Pitch: Kicks, Snares, Toms, ""Shaker"""
		MIDI 22	Crossfade to tertiary Kicks; Pitch: Elec. Snare only
		MIDI 23	"Filter: Kicks, Snares, HiHats, Crashes, Ride, ""Shaker"";"
			Amp LFO: SFX (A6-B6)
		MIDI 24	"EnvCtl: most Kicks, Snares, Toms, ""Shaker"", Elec HiHat"
			LFO Rate: SFX (A6-B6)
		MIDI 25	"(FX3) Mix Level, (Aux) GateRvb Level"
		MIDI 26	"(FX4) Mix Level, GateRvb Level"
		MIDI 27	(FX3) Compressor SmoothTime+MakeUpGain
		MIDI 28	"(FX2) EnvFlt Freq Sweep+Threshold, (FX1) Delay Level"
		MIDI 29	Toggle: Compressor + ChorDelay
58	Lo-Fi Vinyl Kit	MWheel	Pitch for most Needle FX and other SFX
		Data	"Pitch: Kicks, Toms, HiHats"
		MIDI 22	"Pitch: Snares, Crash1"
		MIDI 23	"Assorted Filters: Kick, Toms, Snares,"
			"HiHats, Crashes, Ride (Resonant)"
		MIDI 24	"EnvClt: Kick, Toms, Snares"
		MIDI 25	(FX1) Booth Wet/Dry
		MIDI 26	(Aux) Hall Level
		MIDI 27	"(FX2) Pitcher Wet/Dry, (FX3) LaserVerb Wet/Dry"
		MIDI 28	"(FX2) Pitcher Pitch, (FX3) LaserVerb Delay"
		MIDI 29	Toggle: Pitcher + LaserVerb

ID	Name		
59	Technoo Kit	MWheel	Alternate Kick (B2-C3)
		Data	Pitch: nearly all elements
		MIDI 22	"Filter: Kicks, AuxPerc"
		MIDI 23	"Filter: Snares, Toms, Ride, Crashes, HiHats (A#1-B1)"
		MIDI 24	"EnvCtl: Kicks, Snares (not G#1-A1), Ride, Choke Cym"
		MIDI 25	(FX1) Gated Reverb Wet/Dry
		MIDI 26	Gated Reverb Time
		MIDI 27	(FX1+2) (Aux) LaserVerb Level
		MIDI 28	(FX4) LaserVerb Level
		MIDI 29	Toggle: GateRvb HFDamp+Gate Threshold
60	VAST Sliders 808	MWheel	"AltStart control, Impact on most elements"
		Data	"Pitch: Kicks, Toms"
		MIDI 22	"Pitch: Snares, NoizeToms"
		MIDI 23	"EnvCtl: Kicks, Toms"
		MIDI 24	"EnvCtl: Snares, HiHats, Crash2, NoizeToms"
		MIDI 25	(FX1) Hall Wet/Dry
		MIDI 26	(FX4)- (Aux) Room Level (dry at very top)
		MIDI 27	"Hall Time, Room Decay Time+HFDamp"
		MIDI 28	"(FX2) Flange Wet/Dry+FB, (FX3) 8-Tap Wet/Dry"
		MIDI 29	"Toggle: 8-Tap I/O (Sys), Room Level adj"
61	Industry Set II	MWheel	Filter Resonance (A#4-C5)
		Data	"AltControl on some layers,"
			Pitch on Kick-like elements and some Toms
		MIDI 22	Various Pitch controls on many elements
		MIDI 23	Filters or Modulation Pitch on many elements
		MIDI 24	EnvCtl: assorted kinds of control on many elements
		MIDI 25	"(FX2) Flange Wet/Dry, InEQ: Bass"
		MIDI 26	"(Aux) Hall Level, (FX2) Mix Level"
		MIDI 27	(FX3) DistEQ Wet/Dry+Gain Adjust
		MIDI 28	Distortion Warmth
		MIDI 29	Toggle: RoomType: Hall + Delay
		MPress	Filter Resonance (A#4-C5)
62	General MIDI Kit	MWheel	"Assorted Filters, on most elements"
		Data	"PItch: Kicks (B1, C2), and Toms"
		MIDI 22	"Pitch: Snares (D2, E2), HiHats, Ride, Crash (C#3)"
		MIDI 23	"Pitch: Congas, Timbales, many other elements"
		MIDI 24	"EnvCtl / ASR Amp Env: Kicks (above), Snares (above)"
			"Toms, Crashes, Ride, Triangle, Ding (A#1)"
		MIDI 25	(FX1) Room Wet/Dry
		MIDI 26	Room Rvrb Time
		MIDI 27	"(Aux) Hall Level, (FX1) Mix Level"
		MIDI 28	(FX1) Compressor Ratio+Threshold+Rel Time
		MIDI 29	"Toggle: (FX1) Room+Booth, (Aux) Hall+""Slither Booth"""

ID	Name		
63	Slam'n Drums2 GM	MWheel	"Lowpass Filter and Filter Env Ctrl, many elements"
		Data	"Pitch: Kicks (B1, C2, D6)"
		MIDI 22	"Pitch: Snares (D2, E2), Toms, HiHats"
		MIDI 23	"Filters: Kicks and Snares (above), Toms; HiHat boost"
		MIDI 24	"EnvCtl / ASR Amp Env: Kicks, Snares, Toms"
		MIDI 25	(FX1) Gated Reverb Wet/ Dry+Time+HFDamp
		MIDI 26	(FX2) Sweep Filter Wet/Dry
		MIDI 27	(FX1) FlangeDelay Level (Sys)
		MIDI 28	(FX2) FlangeDelay Level (Sys)
		MIDI 29	Toggle: Sweep Filt dirrection Rise + Fall
64	EleCtroDrumsetGM	MWheel	(FX2) Resonant Filter Freq
		Data	"Filter: Kicks, Toms, assorted other elements"
		MIDI 22	"Pitch: Snares, some Toms, Cymbals,+other elements"
		MIDI 23	"Filter: Snares, Cymbals, HiHats, Synth Boing"
		MIDI 24	EnvCtl: most elements
		MIDI 25	"(FX1) Room Wet/Dry, (FX3) Echo Wet/Dry, (Aux) Hall Wet/ Dry+Level"
		MIDI 26	"Room Time, (Aux) Hall Level"
		MIDI 27	Hall Late Rvrb Time
		MIDI 28	(FX3) Delay Feedback (only a few elements)
		MIDI 29	"Toggle: Room + ResFilt, Delay + Room"
65	SmallKit+Perc MW	MWheel	Cowbell + Shaker Enable
		Data	"Pitch: Kit elements (Kick, Snare, HiHats, Toms, Cymbals)"
		MIDI 22	"Pitch: Congas, Timbales, Agogo, Clave, Cowbell (MW)"
		MIDI 23	"Filters: Cabasas, Tambourines, Clave, Agogo, "
			"Timbales, Kick, Snare, HiHats, Toms, Cowbell (MW)"
		MIDI 24	"Pitch+Filter: Cabasas, Shaker (MW), Tambourine (F#3, F#4)"
		MIDI 25	(FX1+2) Rooms Wet/Dry
		MIDI 26	Rooms' Times
		MIDI 27	"(Aux) Plate Level, (FX4) Mix Level, (FX3) Hall Wet/Dry"
		MIDI 28	Plate Time
		MIDI 29	Toggle: Room + Hall
66	Steel Str Guitar	MWheel	Vibrato
		Data	Layer Enable
		MIDI 22	EnvCtl: Imp
		MIDI 23	EnvCtl: Att+Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Chamber Wet/Dry
		MIDI 26	Chamber Time
		MIDI 27	Chamber HFDamp
		MIDI 28	Comp Ratio
		MIDI 29	Toggle: Pitch I/O
		MPress	Vibrato

ID	Name		
67	12-str Guitar	MWheel	Vibrato
		Data	HFStim EQ
		MIDI 22	EnvCtl: Imp
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	(FX1+2) Room Wet/Dry+(FX1) Time
		MIDI 26	(Aux) Hall Level
		MIDI 27	Hall Time
		MIDI 28	Room + Hall HFDamp
		MIDI 29	Toggle: (fx1+2) Room Reverbs
		MPress	Vibrato
		Soft Pdl	Active
68	Spark Guitar	MWheel	Vibrato
		Data	HFStim adj
		MIDI 22	EnvCtl: Imp+Att
		MIDI 23	EnvCtl: Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	"(fx1) Room Mix, (Aux) Hall Level"
		MIDI 26	Hall PreDelay+Time
		MIDI 27	Delay Mix (sys)
		MIDI 28	Chorus Delay
		MIDI 29	Chorus FB
		MPress	Vibrato
69	Blue Moods	MWheel	"Slight Vibrato, String Balance"
		Data	"String Balance, Gtr Hi Freq Cut"
		MIDI 22	EnvCtl: Imp+Att
		MIDI 23	EnvCtl: Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time+HFDamp
		MIDI 27	"Enhc Lo Mix, Chorus Wet/Dry"
		MIDI 28	"Enhc Hi Mix+Drive, Chorus FB"
		MIDI 29	"Toggle: Enhc + Chorus, Hall + Room"
		MPress	Vibrato
70	SloChrsGtr^Harms	MWheel	Tremolo
		Data	Toggle: SloChorusGuit ^ Harms
		MIDI 22	Para EQ (VAST)
		MIDI 23	"LoPass Res, Chorus (VAST) adj"
		MIDI 24	"EnvCtl: Dec+Rel, Harm(Sin) Oct Shift"
		MIDI 25	(Aux) Hall Level
		MIDI 26	"Hall Time+HFDamp, Chorus Wet/Dry"
		MIDI 27	"Enhc Lo Mix, Chorus FB"
		MIDI 28	Enhc Hi Mix+Drive
		MIDI 29	"Toggle: Enhc + Chorus, Hall + Room"
		MPress	Vibrato

ID	Name		
71	ES335^Abercrmbie	MWheel	Notch Filt Tremolo
		Data	Toggle: ES335 ^ Abercrombie
		MIDI 22	"Para Mid Freq, ^ Notch Freq"
		MIDI 23	"Para Mid Amp (ES335), "
		MIDI 24	"EnvCtl: Att+Rel, ^EnvCtl: Imp+Att"
		MIDI 25	(Aux) Hall Mix
		MIDI 26	"Hall HFDamp, InEQ: Bass+Treb (Abercrmbie)"
		MIDI 27	Chorus Mix
		MIDI 28	Delay Mix
		MIDI 29	Turns off Semi-Tone Pitch Bend
		MPress	Vibrato
		PWheel	Simulates Fretboard Slide (ES335)
72	SliderDistJazzGt	MWheel	Vibrato/Tremolo
		Data	Enables Dist Gtr Layers
		MIDI 22	"Para EQ ^ Hi Freq Stim Drive, Dist EQ"
		MIDI 23	"EnvCtl: Imp, Dist Drive"
		MIDI 24	EnvCtl: Rel
		MIDI 25	"(Aux) FDR Hall Level, Rvb Time"
		MIDI 26	Flange FB
		MIDI 27	Flange Tempo
		MIDI 28	Delay Mix
		MIDI 29	Delay FB
		MPress	"Vibrato, Harmonics Level"
		PWheel	(Dist Layer) +2/-12 Pitch Bend
73	Crunchy Lead	MWheel	Vibrato
		Data	Layer Enable
		MIDI 22	(KDFX)Dist Drive
		MIDI 23	(KDFX)Dist Freq
		MIDI 24	EnvCtl: Dec+Rel
		MIDI 25	"(Aux) FDR Level, Hall Time"
		MIDI 26	Flange FB
		MIDI 27	Flange Tempo
		MIDI 28	Delay Mix
		MIDI 29	Delay FB
		MPress	Layer Balance
74	NooMutes^WahTaur	MWheel	Notch and Para EQ Width
		Data	Toggle: Noo Mutes ^ WahTaur
		MIDI 22	" Notch Freq (Mutes), BandPass Filt LFO Rate (Wah)"
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Att
		MIDI 25	"(FX1) Hall Wet/Dry, (FX3) Room Mix, (Aux) Room Level"
		MIDI 26	"Delay Mix, Chorus Wet/Dry"
		MIDI 27	"Delay FB, Chorus FB"
		MIDI 28	Delay Time
		MIDI 29	Toggle: FDR + Chorus
		MPress	Vibrato
		AttVel	EnvCtl: Decay

ID	Name		
75	CeeTaur^Kotolin	MWheel	Vibrato + EQ
		Data	Toggle: Cee Taur ^ Kotolin
		MIDI 22	EnvCtl: Imp
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	"(Aux) Hall Level, (Fx3) Rvb Time"
		MIDI 26	(Fx2) Phase Wet/Dry
		MIDI 27	"Phase L/R LFO, (Fx3) Flange Mix"
		MIDI 28	Delay Mix
		MIDI 29	"Buss Toggle:, Phaser LFO Rate"
		MPress	Vibrato
76	Liquid T Lead	MWheel	Vibrato
		Data	"EnvCtl: Att, LoPass Freq+Res"
		MIDI 22	"Lopass Freq+Res, Steep Bass Freq"
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	"Hall Time+HFDamp, Chorus FB"
		MIDI 27	"Delay Mix, SRS EQ"
		MIDI 28	"Delay FB, SRS Centerspace"
		MIDI 29	Toggle: CHDelay + SRS
		MPress	"Vibrato, Layer Enable (Harmonics)"
77	Square Lead	MWheel	Vibrato
		Data	"EnvCtl: Att+Dec, LoPass Freq+Res"
		MIDI 22	LoPass Freq+Res
		MIDI 23	InEQ: Bass
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Decay+HFDamp
		MIDI 27	"Flange Wet/Dry, DistDrive adj"
		MIDI 28	"Flange FB+Xcouple+HFDamp, Dist EQ"
		MIDI 29	Toggle: DistFlange I/O
		MPress	"Vibrato, Layer Enable (Harmonics)"
78	AlaZawi Take 2	MWheel	Vibrato
		Data	LoPass Freq+Res
		MIDI 22	LoPass Freq cut
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level+Decay Time
		MIDI 26	Hall PreDelay+HFDamp
		MIDI 27	Chorus Wet/Dry+Pan
		MIDI 28	MDelay Wet/Dry
		MIDI 29	Toggle: Clean + MDelayChorus
		Breath	LoPass Freq+Res adj
		MPress	Vibrato
79	Modulead	MWheel	"Vibrato, Harmonic Balance"
		Data	Filt Freq
		MIDI 22	LoPass Res adj
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Mix
		MIDI 26	Chorus Mix
		MIDI 27	Chorus FB
		MIDI 28	"Phase Center Freq + Notch, Pan Pulse Width"
		MIDI 29	Toggle: Phaser + Panner
		MPress	"Slight Vibrato, LoPass adj"

ID	Name		
80	Polyreal Mini	MWheel	Vibrato
		Data	LoPass Res
		MIDI 22	LoPass Freq
		MIDI 23	"InEQ: Bass, EnvCtl: Att"
		MIDI 24	"InEQ: Treb, EnvCtl: Rel"
		MIDI 25	(Aux) Hall Level+Wet/Dry
		MIDI 26	Hall Decay Time+HFDamp
		MIDI 27	Hall PreDelay
		MIDI 28	Chorus Delay Time+Wet/Dry
		MIDI 29	Delay Mix
		MPress	Vibrato
81	3 Modular Leads	MWheel	Vibrato
		Data	Toggle: Modular Leads
		MIDI 22	"EnvCtl : Imp, HiPass Freq"
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	"(Aux) Hall Level, (FX3) Plate Mix"
		MIDI 26	Hall + Plate PreDelay
		MIDI 27	"Flange Wet/Dry, Delay Mix"
		MIDI 28	"Flange FB, Delay FB"
		MIDI 29	Toggle: Flange + CDR
82	Soul Boy Lead	MWheel	Vibrato
		Data	Filt Freq+Res
		MIDI 22	InEQ: Bass+Treb
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Decay+HFDamp
		MIDI 27	Chorus Mix
		MIDI 28	Delay Mix
		MIDI 29	"Toggle: Delay FB, Chorus adj"
		MPress	Vibrato
83	Fluty Leads	MWheel	Vibrato
		Data	Pitch - Octave Shift
		MIDI 22	InEQ: Bass
		MIDI 23	InEQ: Treb
		MIDI 24	EnvCtl: Rel
		MIDI 25	"(Aux) Hall Level, (FX3) Hall Mix"
		MIDI 26	(Aux) Hall HFDamp+PreDelay
		MIDI 27	Chorus Mix
		MIDI 28	Chorus Depth
		MIDI 29	Toggle: CDR + Room
		MPress	Vibrato
84	Hrmnica^Accordin	MWheel	Vibrato
		Data	Toggle: Hrmnica ^ Accordin
		MIDI 22	Layer Switch
		MIDI 23	"InEQ: Bass, Layer Disable, EnvCtl: Rel"
		MIDI 24	InEQ: Treb
		MIDI 25	"(Fx1) Room Wet/Dry, (Aux) Hall Level"
		MIDI 26	"Room Time, Phase FB"
		MIDI 27	"(Aux) Hall adj, Phase Center Freq+LFODepth"
		MIDI 28	(Aux) Hall HFDamp
		MIDI 29	Toggle: Room + Phaser
		MPress	Vibrato

ID	Name		
85	Trmpts^BrBrass	MWheel	(Brass) HiPass Freq
		Data	Toggle: Trumpets ^ Brass
		MIDI 22	"(Brass) Filt Freq, InEQ: Bass"
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Chamber Level
		MIDI 26	Chamber Time+HFDamp
		MIDI 27	InEQ: Treb
		MIDI 28	Chorus FB
		MIDI 29	Chorus I/O
		MPress	Swell
86	Hip Brass	MWheel	Vibrato
		Data	Freq + Res adj
		MIDI 22	InEQ: Treb
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	"Hall PreDelay Time, (Fx1) Room Time"
		MIDI 27	"(Fx1) Room Wet/Dry+Time, (Fx2) Sweep Filt Res"
		MIDI 28	(Fx2) Sweep Filt LFO Tempo
		MIDI 29	Sweep Filt I/O
		MPress	Swell
87	Brass De' ROM	MWheel	"Filt Freq, Swell"
		Data	"EnvCtl: Imp, Filt Freq + Res"
		MIDI 22	InEQ: Treb + Bass
		MIDI 23	EnvCtl: Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	"(Aux) Hall Level, InEQ: Bass boost"
		MIDI 26	Hall Time+HFDamp
		MIDI 27	Chorus Mix
		MIDI 28	Delay Mix
		MIDI 29	Hall LP Injection+ PreDelay Time
		MPress	"Vibrato, Swell"
88	BigBand^Hornz	MWheel	"Vibrato, LoPass adj (BigBand)"
		Data	Toggle: BigBand ^ Hornz
		MIDI 22	LoPass Freq Cut (Hornz)
		MIDI 23	"LoPass Res (Hornz), EnvCtl: Att"
		MIDI 24	LoPass ASR (Hornz)
		MIDI 25	(Aux) Room Wet/Dry
		MIDI 26	Room Time
		MIDI 27	Room PreDelay
		MIDI 28	Room HFDamp
		MIDI 29	Enhc I/O
		MPress	Vibrato
89	Z-Plane Brass	MWheel	Vibrato
		Data	EnvCtl: Att +Dec
		MIDI 22	BandPass Filt Freq + Width
		MIDI 23	"InEQ: Bass, EnvCtl: Rel"
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time
		MIDI 27	Chorus Mix
		MIDI 28	Delay Mix + Ratio Wet/Dry
		MIDI 29	Delay FB + Hall PreDelay
		MPress	"Vibrato, Slight detune"

ID	Name		
90	OrcBrs^FrenchBone	MWheel	Vibrato
		Data	Toggle: OrcBrs ^ FrenchBone
		MIDI 22	InEQ: Bass
		MIDI 23	"InEQ: Treb, LoPass Freq"
		MIDI 24	EnvCtl: Imp + Rel
		MIDI 25	(Aux) Hall Mix
		MIDI 26	"Hall Time, Mix adj, Pan Rate(Fx3)"
		MIDI 27	Chorus Mix
		MIDI 28	Delay Mix
		MIDI 29	"Hall PreDelay, Pan I/O"
		MPress	"Swell, Vibrato Depth"
91	Brt Saxy Section	MWheel	Vibrato
		Data	"InEQ: Bass, LoPass Freq"
		MIDI 22	InEQ: Treb
		MIDI 23	"EnvCtl: Imp, Att+Dec"
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Room Level
		MIDI 26	"Room Wet/Dry + HFDamp, InEQ: Treb Freq"
		MIDI 27	Dist tube Drive
		MIDI 28	Dist Warmth+Tone
		MIDI 29	"Toggle: Dist+EQ I/O, Room type"
		MPress	Vibrato
92	Brt Saxy/Split	MWheel	Vibrato
		Data	InEQ: Bass+Treb
		MIDI 22	EnvCtl: Imp
		MIDI 23	EnvCtl: Att+Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	Xfade: (fx1) Chmb + (Aux) Hall
		MIDI 26	Chamb + Hall HFDamp
		MIDI 27	Comp smooth Time
		MIDI 28	Comp signal Delay
		MIDI 29	"Toggle: Chamb + Comp, Hall room size"
		MPress	Vibrato
93	Mr. Parker	MWheel	Vibrato
		Data	LoPass Freq
		MIDI 22	LoPass Res
		MIDI 23	LoPass Freq
		MIDI 24	EnvCtl: Att+Rel
		MIDI 25	(Aux) Plate Wet/Dry
		MIDI 26	Plate Time
		MIDI 27	Chorus Mix
		MIDI 28	Delay Mix (sys)
		MIDI 29	"Plate LFO adj, Delay FB"
		MPress	Vibrato
94	Dynasax	MWheel	"Vibrato, LoPass Freq"
		Data	Layer enable
		MIDI 22	"Layer AltCtl, LoPass Freq, Notch Freq, ParaTreb Freq"
		MIDI 23	"Notch Width, LoPass Res, EnvCtl: Imp+Att"
		MIDI 24	EnvCtl: Dec+Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall HFDamp+Decay Time
		MIDI 27	Chorus Mix
		MIDI 28	Delay (sys) Mix
		MIDI 29	Hall PreDelay + room size adj
		MPress	"Vibrato, LoPass Freq+Res, Shape adj"
		ChanSt	"Layer AltCtl, EnvCtl: Rel"
95	DynTrumpet^Miles	MWheel	"swell, Vibrato"
		Data	Toggle: DynTrumpet ^ Miles
		MIDI 22	LoPass Freq+Res
		MIDI 23	"EnvCtl: Imp, InEQ: Bass"
		MIDI 24	"EnvCtl: Rel, InEQ: Treb"
		MIDI 25	"(fx1) Chamb Wet/Dry, (Aux) Room Level"
		MIDI 26	Chamb + Room Times
		MIDI 27	"Chamb + Room HFDamp, Dist Drive"
		MIDI 28	Dist LoPass Freq
		MIDI 29	Toggle: Chamb + Dist
		MPress	Vibrato
96	Wawa Trumpet (MW)	MWheel	"BandPass Freq, Vibrato, RevRvb HFDamp"
		Data	BandPass Width
		MIDI 22	Bandpass LFO Rate
		MIDI 23	"EnvCtl: Imp, InEQ: Bass"
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Wet/ Dry+LateTime+Level
		MIDI 26	(fx2) ReverseRvb Wet/Dry
		MIDI 27	ReverseRvb FB
		MIDI 28	"QFlange Wet/Dry, pan"
		MIDI 29	"QFlange I/O, RevRvb adj, Hall cut"
		MPress	Vibrato
97	Almost Muted (MW)	MWheel	"Vibrato, mute adj"
		Data	LoPass Freq
		MIDI 22	HiPass Freq
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	(fx1) Room Wet/Dry+Time
		MIDI 26	Room HFDamp
		MIDI 27	InEQ: Bass
		MIDI 28	InEQ: Treb
		MIDI 29	EQMorph I/O
		MPress	Vibrato
98	Solo Trombone	MWheel	Vibrato
		Data	LoPass Freq+Res
		MIDI 22	AltControl
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	"(fx1) Chamb Wet/Dry, (Aux) Room Level"
		MIDI 26	Chmb HFDamp
		MIDI 27	InEQ: Treb
		MIDI 28	Dist Drive adj
		MIDI 29	Toggle: Chmb + Dist
		MPress	Vibrato
99	Jazz Band	MW	Tremolo (guitars)
		Data	Toggle: Guitars + Horns
		MIDI 22	Toggle: Band and Drums
		MIDI 23	Tremolo Rate
		MIDI 25	"(Aux) rvb Levels, Wet/Dry"
		MIDI 26	SRS Parameters (guitar layers)
		MIDI 27	(Aux) rvb Times
		MIDI 28	"Early reflection Level, Late Level cut"

ID	Name		
100	W.C.Flute^Winds	Data	Toggle: W.C. Flute ^ Winds; BndPass adj
		MIDI 22	"Sax / square Layers enable, Pan position, BndPass adj"
		MIDI 23	adds chiff (sax / square inactive)
		MIDI 24	EnvCtl: Dec (sax / square)
		MIDI 25	(Aux) Hall Level cut
		MIDI 26	Booth Wet/ Dry+HFDamp+Absorption; Echo Level+Balance
		MIDI 27	"Echo Wet/Dry, FB"
		MIDI 28	Echo Tempo
		MIDI 29	Toggle: Booth + Echo
		MPress	Vibrato
101	Baroque Flute	MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time
		MIDI 27	Hall HFDamp
		MIDI 29	"Toggle: Real Room + Perc Room, InEQ: Treb"
102	Synth Caliopies	MW	Vibrato
		Data	Layer disable(up); LoPass Res
		MIDI 22	BandPass Freq; LoPass Freq
		MIDI 23	"LoPass Freq+Res, Hipass Freq, Treb boost"
		MIDI 24	EnvCtl: Att+Rel
		MIDI 25	"(Aux) Hall Level, Room Wet/ Dry"
		MIDI 26	Phaser FB
		MIDI 27	Phaser LFO Rate
		MIDI 29	"Toggle: Room+Phaser(Layer 1+3), Phaser+CDR(Layer 2+4)"
103	Incan	MW	Vibrato
		Data	HiPass Freq
		MIDI 22	Panner LFO amt+speed
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Plate cut
		MIDI 26	Chorus Level+FB
		MIDI 27	Delay FB
		MIDI 28	Delay Time (4 increments)
		MIDI 29	"Toggle: Echo I/O (sys), Plate Time, Delay Mix"
		MPress	"HiPass Freq LFO amt, Pitch fall"
104	String Orchestra	Data	Toggle: Layers
		MIDI 22	"LoPass Freq cutoff, Bass boost"
		MIDI 23	EnvCtl: Att
		MIDI 25	(FX1) Hall Wet/Dry+PreDelay
		MIDI 26	Hall Time
		MIDI 27	Hall HFDamp
		MIDI 28	(Layer 2) InEQ: Treb cut
		MPress	(Layer 2) volume swell
105	ClassicalStrings	MW	(Layer 3+4) LoPass separation
		Data	"Toggle: Layers1+2, 3+4"
		MIDI 22	(Layer 1+2) LoPass Freq
		MIDI 23	(Layer 3+4) LoPass Freq+Res cut
		MIDI 24	EnvCtl: Att
		MIDI 25	(FX1) Hall Wet/Dry
		MIDI 26	Hall Time
		MIDI 27	Hall HFDamp
		MIDI 28	Hall PreDelay

ID	Name		
106	StSloStr^SilkStr	MW	Layer balance
		Data	Toggle: StSloStr ^ SilkStr
		MIDI 22	"Layer Delay, LoPass Freq+Res "
		MIDI 23	"EnvCtl: Att, LoPass Freq"
		MIDI 24	"EnvCtl: Att+Dec+Rel, LoPass Freq cut"
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time
		MIDI 27	(FX1) Chapel Level
		MIDI 28	Chapel Time
		MPress	"Vibrato, Shaper amt"
107	BigStrgs^FlngStrg	MW	Vibrato ^ Flange Rate
		Data	Toggle: BigStrngs ^ FlngStrg
		MIDI 22	(Layer 1) octave jump
		MIDI 23	EnvCtl: Att
		MIDI 25	(Aux) Hall Level
		MIDI 26	(Aux) Hall Time ^ Phaser CtrFreq
		MIDI 27	Flange FB
		MIDI 28	Flange Rate+L/R phase
		MIDI 29	Toggle: Flange I/O (BigStrgs)
		MPress	"Vibrato, InEQ: Treb"
108	Chamber Ensemble	MW	Chorus Mix
		Data	AllPass Freq
		MIDI 23	EnvCtl: Att
		MIDI 24	InEQ: Bass+Treb
		MIDI 25	"(Aux) Hall Level+PreDelay, (fx1) Hall Mix"
		MIDI 26	Delay Mix+FB
		MIDI 27	Delay Tempo cut
		MIDI 28	(Aux) Hall Time+HFDamp+Injection
		MPress	"Vibrato, AllPass Freq"
109	Psuedo Violin1^2	MW	(Vln1)EnvCtl: Att
		Data	"Toggle: Psuedo Violin1 ^ 2, (Vln2) EnvCtl: Att"
		MIDI 22	(Vln1) LoPass Freq cut
		MIDI 23	(Vln2) LoPass Freq cut
		MIDI 24	(Vln1) Shaper Level
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall LateRvbTime+HFDamp
		MIDI 27	(FX1) Room Wet/Dry
		MPress	Vibrato
110	Slo Solo Cello	MW	LoPass sep (expRession / dynamic ctl)
		Data	LoPass Freq cut+Res cut
		MIDI 22	EnvCtl: Att+Dec
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall LateRvbTime+HFDamp
		MIDI 27	(FX1) Room Wet/Dry
		MIDI 28	Room HFDamp
		MIDI 29	"Toggle: Room type, Hall Wet/ Dry"
		MPress	Vibrato

ID	Name		
111	Horn&Flute w/Str	MW	"Vibrato, LoPass sep (expression / dynamic ctl)"
		Data	Toggle: Horn ^ Solo String
		MIDI 22	LoPass Freq+Res cut
		MIDI 23	Ens Strings Vol cut
		MIDI 24	Ens Strings EnvCtl: Att
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time
		MIDI 27	(FX1) Chapel Wet/Dry
		MIDI 28	Chapel Time
		MIDI 29	"Toggle: (Layer 3+4) Chapel+Hall, (Layer 1) Hall+Chapel"
		MPress	Ens Strings Vibrato
		SostPd	Toggle: Solo Strg I/O
112	DynOrch^WTellOrc	MW	string and brass balance
		Data	Toggle: DynOrch ^ WTellOrch
		MIDI 22	"ParaMid and LoPass Freq, Shaper Drive"
		MIDI 23	"Shaper amt, LoPass Freq"
		MIDI 25	(Aux) Hall Level cut
		MIDI 26	Chapel + Hall Times
		MIDI 29	Toggle: Chapel/Hall + Hall/ Room
		MPress	"(DynOrch) Volume swell, shaper amt"
		SostPd	"Layer enable, Room Time"
113	Touchy Orchestra	MW	"Bone cut, LoPass Freq(Lo Strings)"
		Data	Hi String cut
		MIDI 22	Lo String ^ Hi Flute
		MIDI 23	Timpani cut
		MIDI 24	Loud Horn ^ Soft Horn
		MIDI 25	"(Aux) Hall Level, Chapel + Rooms Wet/Dry"
		MIDI 26	Chapel Time
		MIDI 29	"Toggle: Chapel+Room, Hall+Chapel"
		MPress	Hi Flute Vibrato Rate
114	Synth Strings	MW	"Vibrato, modulation"
		Data	Toggle: Layer 1 ^ Layer 3
		MIDI 22	Layer 1 up p5th ^ Layer 3 up 8ve
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Imp+Rel
		MIDI 25	(Aux) Plate Level
		MIDI 26	"Chorus Wet/Dry, Dist Drive"
		MIDI 27	Chorus FB
		MIDI 28	Dist Bass+Treb tone
		MIDI 29	Toggle: Chorus + Distortion
		MPress	"Vibrato, modulation"
115	Mellostr^ShineOn	MW	Vibrato
		Data	Toggle: Mellostr ^ ShineOn
		MIDI 22	LoPass+BandPass Freq+Width
		MIDI 23	"EnvCtl: Att, LoPass Res"
		MIDI 24	EnvCtl: Rel
		MIDI 25	"(Aux) Room Level, Hall absorption"
		MIDI 26	"Filt Res, Chorus FB"
		MIDI 27	"Filt Freq, Chorus Rate"
		MIDI 28	"Filt Vibrato, Delay Mix"
		MIDI 29	Toggle: Res Filt + ChorDelay (Mellostr only)
		MPress	"Vibrato, HiPass Freq"

ID	Name		
116	Mellotron (MW)	MW	"3-way Toggle: Ens Strg, Solo Strg(dwn 8ve), Flute"
		Data	Octave jump
		MIDI 22	LoPass Freq; ParaTreb Freq ; HiFreqStim Freq
		MIDI 23	"Dist Drv, Xfade dpth; ParaTreb dpth; HFStim Drv"
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time
		MIDI 27	Room Level
		MIDI 28	Room Time
		MPress	Vibrato
117	RaveStrg^Solina	MW	"Vibrato, Layer detune(Sol)"
		Data	Toggle: RaveStrg ^ Solina
		MIDI 22	EnvCtl: Att+Rel
		MIDI 23	"EnvCtl: Dec ^ Ptch mod, Notch LFO Rate"
		MIDI 24	"Flange Mix, Spin Wet/Dry"
		MIDI 25	(Aux) Room Level
		MIDI 26	Spin Pitcher Mix ^ MovDelay Wet/Dry
		MIDI 27	Spin Pitcher Weights
		MIDI 28	Spin Pitcher ptch (rapid echo Rate)
		MIDI 29	"Toggle: Spin I/O, Room HFDamp+Time"
		MPress	Vibrato
118	Choir Strings	Data	LoPass Freq cut+Res (string)
		MIDI 22	LoPass Freq cut (vox)
		MIDI 23	"Layer detune, LoPass Res"
		MIDI 24	Panner Width
		MIDI 25	(Aux) Room Level
		MIDI 26	(Aux) Room Time
		MIDI 27	Flange Level
		MIDI 28	Flange Tempo
		MIDI 29	"Toggle: Room + Flange (string), ChHall + Hall (vox)"
119	CathdrVox^8veVox	MW	"Vibrato+Rate (CathV), Sin Tremolo Rate (8veV)"
		Data	Toggle: CathedralVox ^ 8veVox
		MIDI 22	"EnvCtl: Att, LoPass Freq, Xfade Lo/Hi Vox(8veV)"
		MIDI 23	"EnvCtl: Rel, Panner pos, 8ve jump(CathV)"
		MIDI 24	InEQ: Treb cut
		MIDI 25	(Aux) Hall Level
		MIDI 26	(Aux) Hall Time+build Time
		MIDI 27	Delay Mix+FB
		MIDI 28	Flange Mix+FB
		MPress	"Vibrato+Rate (CathV), Sin Tremolo Rate (8veV)"
120	Mixed Choir	MW	Vibrato+Rate
		Data	Layer XFade
		MIDI 22	"EnvCtl: Rel, Notch + ParaTreb Freq"
		MIDI 23	"InEQ: Bass, ParaTreb, Notch Width"
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level
		MIDI 26	Room Wet/Dry
		MIDI 27	Room Time
		MIDI 28	Infinite Decay on Keydown
		MIDI 29	Infinite Decay
		MPress	Vibrato+Rate

ID	Name		
121	NUChoir^SpaceVox	MW	Vibrato
		Data	Toggle: NUChoir ^ SpaceVox
		MIDI 22	"EnvCtl: Att, Shaper amt ^ LoPass Freq"
		MIDI 23	EnvCtl: Rel
		MIDI 24	LoPass Freq (SpaceVox)
		MIDI 25	(Aux) Hall: Level+buildTime+DecayTime+PreDelay+HFDamp
		MIDI 26	(Aux) Hall Bass gain
		MIDI 27	InEQ: Bass
		MIDI 28	InEQ: Treb
		MIDI 29	Toggle: (Aux) Hall I/O (SpaceVox)
		MPress	Vibrato
122	Marimbae^Vibeish	MW	"Layer 2 alt Toggle, EnvCtl: Rel, Tremolo, Panner pos"
		Data	Toggle: Marimbae ^ Vibish
		MIDI 22	"Xfde: Layer 1+2, Layer detune, LP / HPass freq, HFStim Drive"
		MIDI 23	"Layer 2 alt Toggle, EnvCtl: Rel"
		MIDI 24	EnvCtl: Imp+Dec
		MIDI 25	"(Aux) Hall Level, Room Wet/ Dry"
		MIDI 26	Hall+Room Times
		MIDI 29	Toggle: Room + Compressor/ Hall ^ Room I/O
123	SlidersIn Africa	MW	"Layer 2 Xfade, Panner Pos"
		Data	LoPass Freq cut
		MIDI 22	HiPass Freq
		MIDI 23	Panner LFO amt+Pos
		MIDI 24	Pitch drop
		MIDI 25	Delay Time
		MIDI 26	Delay FB
		MIDI 27	Laser Contour
		MIDI 28	Laser Spacing
		MIDI 29	(Aux) Room Time adj
124	MalletFlts Split	MW	Pitch up (Layer1+6)
		Data	Layer5 Enable
		MIDI 22	"LoPass Freq+Res, Shape mod (Layer 4), ParaTreb boost"
		MIDI 23	"HiPass Freq (Layer3), Shaper amt, Notch Freq (Layer 4), Panner Pos(Layer 3)"
		MIDI 25	(Aux) Space Rvb Level+Mix+Time
		MIDI 26	Delay Mix
		MIDI 27	Delay Time
		MIDI 29	"Toggle: CDR + Chorus, (Layers 1, 2, 6)"
		MPress	Notch Freq
125	Malletoo	MW	"Pitch modulation, Vibrato"
		Data	"Pitch mod LFO speed, LoPass Freq (Layer 3)"
		MIDI 22	LoPass Res cut (Layer 3)
		MIDI 23	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level+Time+HFDamp
		MIDI 26	(Aux) Delay Mix cut
		MIDI 27	Flange Wet/Dry
		MIDI 28	Flange FB
		MIDI 29	Toggle: Flange I/O
		MPress	Vibrato

ID	Name		
126	Highlandistic	MW	(Aux) Hall Infinite Decay Vibrato
		Data	EnvCtl: Rel
		MIDI 22	ParaTreb Freq
		MIDI 23	"InEQ: Bass, Notch+HiPass Freq"
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level
		MIDI 26	"Hall Time+HFDamp, Chorus Mix"
		MIDI 27	Delay Mix+FB
		MIDI 28	Chorus Level cut
		MIDI 29	Toggle: Infinite Decay I/O
		MPress	Vibrato
127	Buzz Kill	MW	Pitch modulation
		Data	LoPass Freq
		MIDI 22	"LoPass Res cut, Dist Drive cut"
		MIDI 23	"EnvCtl: Att, Flange LFO"
		MIDI 24	"EnvCtl: Rel, Flange L/R phase"
		MIDI 25	Flange Delay Tempo
		MIDI 26	Flange FB
		MIDI 27	(Aux) CDR Level cut
		MIDI 28	(Aux) Delay Mix
		MIDI 29	(Aux) Hall Wet/Dry+Time adj
		MPress	LoPass Freq
128	Harshey's	MW	Pitch modulation
		Data	LoPass Freq
		MIDI 22	"LoPass Res cut, Dist Drive cut"
		MIDI 23	"EnvCtl: Att, Flange LFO"
		MIDI 24	"EnvCtl: Rel, Flange L/R phase"
		MIDI 25	"Flange Wet/Dry, (Aux) Delay Tempo"
		MIDI 26	Flange FB
		MIDI 27	(Aux) CDR Level
		MIDI 28	(Aux) Delay Mix
		MIDI 29	(Aux) Hall Wet/Dry+Time adj
		MPress	LoPass Freq
129	Spunter	MW	Pitch octave jump
		Data	"EnvCtl: Dec, Rel"
		MIDI 22	Pitch octave+jump
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level+Mix
		MIDI 26	Hall Time
		MIDI 27	(Aux) Delay Mix+Dry Balance
		MIDI 28	Distortion Wet/ Dry+Warmth+HiPass
130	Crumper	MW	Vibrato
		Data	EnvCtl: Att+Rel
		MIDI 22	DSP Xfade: SW+SHP
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Mix+Time+PreDelay
		MIDI 26	(Aux) Delay Mix
		MIDI 27	Dist Drive
		MIDI 28	Dist Warmth
		MIDI 29	Toggle: Delay I/O
		MPress	Vibrato

ID	Name		
131	Zeek	MW	Pitch modulation
		Data	LoPass Freq cutoff
		MIDI 22	"LoPass env amt, 8ve jump"
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	(FX2) Room Wet/Dry cut
		MIDI 26	Room Time
		MIDI 27	Dist Wet/Dry
		MIDI 28	(Aux) GrphEQ I/O
		MIDI 29	Toggle: AClock Amp modulation I/O
132	Elephantiasis	MW	HiPass Freq
		Data	HiPass Freq
		MIDI 22	Pitch
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) EQ 1k Level
		MIDI 26	(Aux) EQ 2k Level
		MIDI 27	(Aux) EQ 4k Level
		MIDI 28	(Aux) EQ 8k Level
		MIDI 29	(Aux) Level cut
133	SoftMatrix12 5th	MW	Vibrato
		Data	"LoPass Freq, EnvCtl: Att+Dec"
		MIDI 22	Pitch drop
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Plate Level
		MIDI 26	Plate HFDamp+Decay Time
		MIDI 27	Echo Wet/Dry
		MIDI 28	Echo FB
		MIDI 29	"Toggle: Echo I/O, Plate room size adj"
		GAttVel	Echo HFDamp
		MPress	Vibrato
134	TchRezoid^Hungry	MW	"Vibrato, Phaser Ctr Freq"
		Data	Toggle: TchRezoid (EnvCtl: Att+Dec) ^ Hungry (EnvCtl: Rel)
		MIDI 22	"HiPass Res, Wrap amt ^ LoPass Res, "
			"Saw+Shp 8ve jump, Xfade: triangle+saw"
		MIDI 23	HiPass sep ^ EnvCtl: Att
		MIDI 24	EnvCtl: Rel ^ EnvCtl: Imp
		MIDI 25	AstralTaps Wet/Dry+FB
		MIDI 26	(Aux) Phaser Level
		MIDI 27	"Phaser Notch, BandPass adj"
		MIDI 28	AstralTaps Tempo
		MPress	Vibrato
135	Matchstik^NUDigi	MW	"Vibrato, Phaser Ctr Freq"
		Data	Toggle: MatchStik ^ NUDigi
		MIDI 24	(Aux) Phaser Level+FB cut
		MIDI 25	(Aux) Phaser Notch/ BP
		MIDI 26	AstralTaps Wet/Dry+FB
		MIDI 27	AstralTaps FB image
		MIDI 28	AstralTaps Tempo
		MPress	Vibrato

ID	Name		
136	Peppers Digiclub	MW	"Pitch chng+mod, LFO Depth(VAST+(Aux) Hall)"
		Data	LoPass Freq
		MIDI 22	"Pitch jump, Pitch LFO Rate Toggle"
		MIDI 23	EnvCtl: Att
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level (MIDI 29 out)
		MIDI 26	(Aux) Hall Time
		MIDI 27	Flange Wet/Dry
		MIDI 28	Flange Feedback
		MIDI 29	"(Aux) Hall Level+Mix+LFO Depth+Rate, Flange Dry cut"
		MPress	Pitch changes
137	Frog^LowNoteTalk	MW	Vibrato
		Data	Xfade: Frog ^ LowNoteTalk
		MIDI 22	"Pitch Drop, Wrap amt"
		MIDI 23	ShpOsc Depth ^ Wrap amt
		MIDI 24	BandPass+AllPass Freq
		MIDI 25	"EQ Morph ""A"" Freq scale"
		MIDI 26	"EQ Morph ""B"" Freq scale"
		MIDI 27	Echo Wet/Dry
		MIDI 28	Echo FB
		MIDI 29	Toggle: EQ Morph + Echo
		MPress	Vibrato
138	Oink Doink	MW	Vibrato
		Data	EnvCtl: Att+Dec
		MIDI 22	HiPass Res cut
		MIDI 23	"InEQ: Bass, HiPass separation"
		MIDI 24	"InEQ: Treb, EnvCtl: Imp"
		MIDI 25	(Aux) Hall Level
		MIDI 26	(Aux) Delay Level
		MIDI 27	Laser spacing
		MIDI 28	Laser Contour
		MIDI 29	Toggle: Laser I/O
		MPress	Vibrato
		LgeRbn	Laser Delay Coarse
139	Wheepy Dood	MW	Vibrato
		Data	LoPass Freq
		MIDI 23	EnvCtl: Att+Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	Chorus Mix+Delay
		MIDI 26	Chorus Depth+Rate
		MIDI 27	Chorus FB
		MIDI 28	Delay Mix+FB
		MPress	Vibrato
140	Funk O Matic	MW	"Vibrato, Vibrato Rate"
		Data	LoPass Freq
		MIDI 22	"Shaper amt, LoPass Freq cut"
		MIDI 23	Dist Drive
		MIDI 24	(Layer 1+3) 8ve drop
		MIDI 25	Env Filt thReshold
		MIDI 26	Env Filt min Freq
		MIDI 27	(Aux) Sweep Filt Wet/Dry
		MIDI 28	(Aux) Sweep Filt min Freq
		MIDI 29	Toggle: Env Filt - BandPass and HiPass
		MPress	"Vibrato, Layer detune"

ID	Name		
141	Pulsemonster5ths	MW	Vibrato
		Data	"PWM Width, LoPass Freq, Dist Drv cut"
			EnvCtl: Att+Dec+Rel
		MIDI 22	Dist Drv cut
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level
		MIDI 26	"(Aux) Hall PreDelay, Decay Time, HFDamp, Bass gain"
		MIDI 27	Chorus Depth
		MIDI 28	Delay Mix+FB
		MPress	Vibrato
142	OB Pad^OB Brass	MW	Vibrato
		Data	Toggle: OBPad ^ OB Brass
		MIDI 22	"LoPass Freq, EnvCtl: Att+Rel (pad)"
		MIDI 23	LoPass Res ^ LoPass Freq
		MIDI 24	EnvCtl: Imp (brass)
		MIDI 25	(Aux) Plate Level+Time
		MIDI 26	"Enhc Lo Drive+Mix, Chorus Wet/Dry "
		MIDI 27	"Enhc Mid Drive, Mid Mix"
		MIDI 28	"Enhc Hi Drive, Hi Mix, InEQ: Treb"
		MIDI 29	Toggle: Enhancer + Chorus
		MPress	Vibrato
143	Detooner^BigPMW	MW	Vibrato
		Data	Toggle: Detooner ^ BigPMW
		MIDI 22	"P5th jump ^ LoPass Freq, EnvCtl: Att+Rel"
		MIDI 23	"Notch Freq ^ Dist drv, EnvCtl: Imp"
		MIDI 24	"PWM Width, Dist drv"
		MIDI 25	(Aux) Laser Level
		MIDI 26	(Aux) Laser contour+FB
		MIDI 27	"Flange FB+L/R phase, Phaser Ctr Freq"
		MIDI 28	"Flange Wet/Dry cut, Phaser Wet/Dry"
		MIDI 29	Toggle: Flange + Phaser
		MPress	Vibrato
144	TeknoBallCrusher	MW	Vibrato
		Data	"EnvCtl: Att, Notch Freq"
		MIDI 22	saw 8ve jump (Layer 1)
		MIDI 23	EnvCtl: Impact
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Room Level
		MIDI 26	Chorus Wet/Dry; Dist Drive cut
		MIDI 27	Chorus Rate; Dist warmth cut
		MIDI 28	Chorus FB; Dist cab LoPass
		MIDI 29	Toggle: Chorus + Distortion
		MPress	Vibrato
145	VTrig SquareHead	MW	Vibrato
		Data	BandPass Freq Env amt+Width
		MIDI 23	"EnvCtl: Imp, Saw Pitch, InEQ: Bass "
		MIDI 24	"EnvCtl: Rel, InEQ: Treb"
		MIDI 25	(Aux) Delay Mix
		MIDI 26	(Aux) Chorus Mix
		MIDI 27	Phaser FB cut
		MIDI 28	Phaser Ctr Freq
		MIDI 29	(Aux) Chorus Delay cut
		MPress	Vibrato

ID	Name		
146	Razor Saw	MW	Vibrato
		Data	"LoPass LFO Rate, Shaper amt, EnvCtl: Att+Dec "
		MIDI 22	EnvCtl: Rel
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level+PreDelay+Time+HFDamp
		MIDI 26	Delay FB+Mix
		MIDI 27	Chorus Depth+Rate
		MIDI 28	Chorus FB
		MIDI 29	Toggle: Delay I/O
		MPress	Vibrato
147	Razorip Buzzsaw	MW	Vibrato
		Data	"EnvCtl: Rel, HiPass Freq"
		MIDI 22	Sine Pitch
		MIDI 23	Saw+Sine Pitch
		MIDI 24	InEQ: Bass+Treb
		MIDI 25	(Aux) Hall Time+HFDamp
		MIDI 26	(Aux) Delay Mix
		MIDI 27	Phaser FB cut
		MIDI 28	Phaser Ctr Freq+LFO Rate
		MIDI 29	(Aux) Chorus Delay adj
		MPress	"Vibrato, Phaser LFO Rate"
148	Pulse Pass	MW	Vibrato
		Data	HiPass Freq
		MIDI 22	PWM Width
		MIDI 23	EnvCtl: Imp
		MIDI 24	"EnvCtl: Att+Rel, Dist drv cut"
		MIDI 25	"(Aux) Hall Level+Time, Delay HFDamp"
		MIDI 26	Delay Mix
		MIDI 27	Chorus Rate+Depth
		MIDI 28	Chorus FB
		MIDI 29	(Aux) Hall PreDelay adj
		MPress	Vibrato
149	Nordic Square	MW	Vibrato
		Data	Lo+Hi Pass Freq cut
		MIDI 22	EnvCtl: Att
		MIDI 23	EnvCtl: Imp
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level; Flange/ Dry>Delay Mix
		MIDI 26	"Chorus Mix, Depth+Delay"
		MIDI 27	Chorus Rate+FB
		MIDI 28	Delay Mix+FB
		MIDI 29	Toggle: ChorusDelay/(Aux) Hall + Flange/Delay
		MPress	Vibrato
150	Rezonant Snob	MW	Vibrato
		Data	Filt LFO
		MIDI 22	Filt Res
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Wet/Dry
		MIDI 26	Hall Time
		MIDI 27	Hall preDelay+HFDamp
		MIDI 28	Delay Mix
		MIDI 29	Toggle: Delay Time cut+HFDamp adj
		MPress	"Vibrato, Filt Freq"

ID	Name		
151	Borfin'Jumps^RPT	MW	"EnvCtl: Att+Dec, Rel"
		Data	Toggle: Borfin'Jumps ^ RPT
		MIDI 22	Pitch jump
		MIDI 23	"LoPass Res, "
		MIDI 24	"KDFX InEQ: Treb, Env Ctl (Impact), Amp modulation"
		MIDI 25	"(Aux) rvb Level + Time, PreDelay Time"
		MIDI 26	"Delay Mix, Chorus Delay Time"
		MIDI 27	Phaser FB
		MIDI 28	Phaser LFO Rate
		MIDI 29	Phaser Notch/BandPass balance
		MPress	Phaser LFO Rate
152	Elektro Funk	MW	Modulation
		Data	"Synch slave de-tune, Notch Freq"
		MIDI 22	Synch Master de-tune
		MIDI 23	EnvCtl Att
		MIDI 24	EnvCtl Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	"Hall Time, size"
		MIDI 27	Vibrato Phaser Wet/Dry
		MIDI 28	"Vibrato Phaser Width, InEQ: Treb"
		MIDI 29	Toggle: Echo on
		MPress	
153	Acid	MW	Pitch up
		Data	HiPass Freq
		MIDI 22	LoPass Res Freq
		MIDI 23	LoPass Res env
		MIDI 24	EnvCtl: Rel
		MIDI 25	Shaper Mid amount
		MIDI 26	Delay Level
		MIDI 27	Delay FB
		MIDI 28	Delay Time
		MIDI 29	Toggle: (Aux) Hall Level boost
154	Rancid	MW	Pitch adj
		Data	HiPass Freq
		MIDI 22	LoPass Res Freq
		MIDI 23	LoPass Res env amt
		MIDI 24	EnvCtl: Rel
		MIDI 25	Shaper Mid amount
		MIDI 26	Echo Level
		MIDI 27	Echo FB
		MIDI 28	Echo Time
		MIDI 29	Toggle: (Aux) Hall Level boost
155	Fuse	MW	System-Synched modulation
		Data	LoPass Freq
		MIDI 22	Lo pass Res cut
		MIDI 23	Pitch drop ASR
		MIDI 24	EnvCtl: Rel
		MIDI 25	Env Filt amt
		MIDI 26	Env Filt min Freq
		MIDI 27	InEQ: Bass
		MIDI 28	InEQ: Treb
		MIDI 29	Toggle: Sweep Filt I/O
		MPress	Gain boost

ID	Name		
156	Squwee Monosync	MW	Vibrato
		Data	Layer enable
		MIDI 22	EnvCtl: Att
		MIDI 23	EnvCtl: Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Booth Level
		MIDI 26	"Gated rvb Wet/Dry cut, Flange Level cut"
		MIDI 27	"Gated rvb size scale, Echo Level"
		MIDI 28	"Echo HFDamp, Echo Level + FB "
		MIDI 29	Toggle: Gated rvb + FlangeEcho
		MPress	Vibrato
157	Generic Ravelead	MW	Pitch mod
		Data	"LoPass Freq cut, EnvCtl: Att+Dec"
		MIDI 22	"LoPass Res, HiPass Freq"
		MIDI 23	LoPass Res cut
		MIDI 24	"Samp alt start, EnvCtl: Imp"
		MIDI 25	(Aux) Booth Level
		MIDI 26	"Echo Wet/Dry, LFO Filt pulse Width"
		MIDI 27	"Echo FB, LFO Filt smooth"
		MIDI 28	"Echo HFDamp, LFO Filt Res"
		MIDI 29	Toggle: Echo + LFO Filt
		MPress	Vibrato
158	Notch Sinker	MW	Vibrato
		Data	EnvCtl: Att+Dec
		MIDI 22	SyncM Pitch adj
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Booth Level
		MIDI 26	"Shaper Wet/Dry cut, Reverse rvb Wet/Dry"
		MIDI 27	"Shaper amount, Reverse rvb envelope"
		MIDI 28	"Reverse rvb length, Delay length"
		MIDI 29	"Toggle: shaper + reverse rvb, (Aux) Booth type+Time"
		MPress	Vibrato
159	Grungesync Sweep	MW	Vibrato
		Data	EnvCtl: Att+Dec
		MIDI 22	Layer LoPass Freq cut
		MIDI 23	Saw wave Pitch jump
		MIDI 24	EnvCtl: Imp
		MIDI 25	(Aux) Hall Level
		MIDI 26	Flange Level
		MIDI 27	"Shaper Level, Quantize Wet/ Dry"
		MIDI 28	Shaper amount
		MIDI 29	"Toggle: ShaperFlange + QuantizeFlange, (Aux) Hall Time"
		MPress	Vibrato

ID	Name		
160	Rezzysaws	MW	Vibrato
		Data	LoPass Freq
		MIDI 22	EnvCtl: Att
		MIDI 23	Saw wave Pitch adj
		MIDI 24	EnvCtl: Imp
		MIDI 25	"(Aux) Hall Level, Wet/Dry"
		MIDI 26	"EnvFilt thReshold, Chorus Level"
		MIDI 27	"EnvFilt sweep, Delay Level"
		MIDI 28	"EnvFilt min Freq, Delay Time"
		MIDI 29	Toggle: EnvFilt + ChorDelay
		MPress	Vibrato
161	Prophet Sync	MW	"Pitch adj, Vibrato"
		Data	"LoPass Freq + Res, Sync slave Pitch, Layer EnvCtl: Rel"
		MIDI 22	Layer Pitch adj
		MIDI 23	InEQ: Treb
		MIDI 24	EnvCtl: Att + Imp
		MIDI 25	(Aux) rvb Level + HFDamp
		MIDI 26	"FX2 Flange Level, FX3 Flange Wet/Dry"
		MIDI 27	"Shaper Level, Quantize Wet/Dry"
		MIDI 28	Shaper amount
		MIDI 29	Toggle: FlangeShaper + QuantizeFlange
		MPress	Pitch mod
162	Glider	MW	"LoPass LFO Rate, Vibrato"
		Data	LoPass Freq LFO amt
		MIDI 22	LoPass Res
		MIDI 23	Notch Freq
		MIDI 24	"Flange FB, EnvCtl: Imp"
		MIDI 25	(Aux) LaserDelay Level
		MIDI 26	"Flange Wet/Dry, Phaser Wet/Dry"
		MIDI 27	LaserDelay Time
		MIDI 28	LaserDelay FB
		MIDI 29	"Toggle: Flange + Phaser, LaserDelay adj"
		MPress	Vibrato
163	AlaskaGlide (MW)	MW	Toggle: Alaska + Glide
		Data	EnvCtl: Imp
		MIDI 22	EnvCtl: Att
		MIDI 23	EnvCtl: Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Levels
		MIDI 26	FDR Wet/Dry
		MIDI 27	InEQ: Bass
		MIDI 28	InEQ: Treb
		MIDI 29	FlgDelayrvb I/O
		MPress	"Vibrato, Layer detune, LoPass Freq, Flange XCurs + FB"
164	RaveClassic^Braz	MW	Vibrato
		Data	Toggle: RaveClassic + Braz
		MIDI 22	"EnvCtl: Imp ^ NotchFilt Freq, EnvCtl: Imp"
		MIDI 23	EnvCtl: Dec ^ EnvCtl: Att+Dec
		MIDI 24	EnvCtl: Rel ^ EnvCtl: Rel
		MIDI 25	"Hall Mix + Time, Delay Mix"
		MIDI 26	Panner LFO Rate
		MIDI 27	Pan image Width
		MIDI 28	Delay Tempo
		MIDI 29	Echo I/O
		MPress	Vibrato
165	Meditator^SloEns	MW	"Vibrato, LoPass Res"
		Data	Toggle: Meditator + SloEns
		MIDI 22	"LoPass Freq + Res, HFstim adj, Layer Pitch adj"
		MIDI 23	"BandPass Freq, Layer Pitch adj (SloEns)"
		MIDI 24	"Layer Shape adj, EnvCtl: Imp"
		MIDI 25	(Aux) Hall Level + Decay Time ^ Miniverb Level
		MIDI 26	Flang Wet/Dry ^ Minivrb Time + PreDelay
		MIDI 27	Flange FB
		MIDI 28	Delay FB
		MIDI 29	Toggle: Flange + CDR
		MPress	Vibrato
		AttVel	EnvCtl: Rel
		MIDI 70	"AltCtl, EnvCtl: Imp"
166	Tranquil Picked	MW	"Vibrato, Notch Freq env amt"
		Data	"Layer LoPass Freq + Xfade, AllPass Freq, Notch Width, EnvCtl: Rel"
		MIDI 22	"Layer HiPass Freq, Pan adj, AllPass Freq"
		MIDI 23	"Layer Pitch adj, AllPass adj"
		MIDI 24	"Layer Pitch env, EnvCtl: Att"
		MIDI 25	(Aux) Echo Level
		MIDI 26	Flange Wet/Dry
		MIDI 27	Laser coarse Delay
		MIDI 28	Laser contour
		MIDI 29	Toggle: Flange + LaserVrb
		MPress	Vibrato
167	The Chaser	MWheel	Vibrato
		Data	"LoPass Freq, EnvCtl: Att+Dec+Rel"
		MIDI 22	LoPass Freq+Res
		MIDI 23	"HFstim adj, Dist Drive"
		MIDI 24	PWM Width
		MIDI 25	"Rooms Wet/Dry (FX1,FX2)"
		MIDI 26	Rooms HFDamp+Time
		MIDI 27	(Aux) Hall Level
		MIDI 28	Hall HFDamp
		MIDI 29	Toggle: Rooms
		AttVel	AltCtl
		KeyNum	"HFstim LFO, BandPass Width, Notch Freq"
		MPress	Vibrato
168	Enterprize^MTree	MWheel	"Vibrato, Tremolo"
		Data	Toggle: Enterprize ^ MTree
		MIDI 22	"Pitch jump, HFStim ^ EnvCtl: Att+Dec"
		MIDI 23	"HiPass Freq, Dist Drive (VAST)"
		MIDI 24	"DSP XFade, Pitch adj, EnvCtl: Rel"
		MIDI 25	(Aux) Acid Room Level
		MIDI 26	"Acid dry Level cut, Dist Drive adj ^ LasrVrb Wet/Dry"
		MIDI 27	Dist warmth ^ LasrVrb Delay Time
		MIDI 28	Dist Freq adj ^ LasrVrb contour
		MIDI 29	Distortion I/O
		MPress	"Vibrato, Tremolo"
		AttVel	EnvCtl: Rel

ID	Name		
169	Orgazmica	MWheel	Layer detune
		Data	LoPass Freq cut
		MIDI 22	Pitch octave adj
		MIDI 23	"EnvCtl: Att, HiPass Freq"
		MIDI 24	"EnvCtl: Rel, LoPass Res"
		MIDI 25	"(Aux) Booth Level, (FX1) Room Wet/Dry"
		MIDI 26	Hall HFDamp+Absorption+Time
		MIDI 27	Tremolos
		MIDI 28	Flange FB + Wet/Dry
		MIDI 29	Toggle: (FX1) Room I/O
170	Universal	MWheel	Vibrato
		Data	Pitch adj
		MIDI 22	Amp gain
		MIDI 23	Pitch adj
		MIDI 24	"Saw detune, Pan adj, Square Level "
		MIDI 25	(Aux) Hall Level
		MIDI 26	Flange FB
		MIDI 27	Flange LFO Tempo
		MIDI 28	Flange Xcouple
		MIDI 29	"Toggle: Room + Flange, Hall Decay Time"
		MPress	Layer Vibrato + boost
171	Asian Digital	MWheel	"Vibrato, Shaper Level"
		Data	EnvCtl: Att+Dec
		MIDI 22	SW+SHP Pitch+pan+Level
		MIDI 23	EnvCtl: Imp+Dec
		MIDI 24	(Aux) Delay FB
		MIDI 25	(Aux) Hall Mix+Time
		MIDI 26	(Aux) Chorus FB
		MIDI 27	Flange Wet/Dry
		MIDI 28	Flange FB+phase
		MIDI 29	Toggle: Flange I/O
		MPress	"Vibrato, Shaper Level"
		KeyNum	Pos bal LFO
172	Wheel Wave Sweep	MWheel	"Layer Xfade, vib, Pan LFO, BandPass Width"
		Data	"BandPass Freq+Width, LoPass + HiPass adj"
		MIDI 22	"LoPass Res, HiPass Res, Pan adj"
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time+HFDamp
		MIDI 27	Flange FB Level
		MIDI 28	Flange LFO Tempo
		MIDI 29	Toggle: Flange I/O
		MPress	Vibrato
173	Arystal^InTheAir	MWheel	Vibrato
		Data	Toggle: Arystal ^ InTheAir
		MIDI 22	Layer Pitch adj ^ LoPass adj
		MIDI 23	"LoPass Freq ^ Saw Pitch, Layer detune"
		MIDI 24	"Layer Pitch adj, Layer Xfade"
		MIDI 25	(Aux) Hall Level+Time
		MIDI 26	Chorus Wet/Dry
		MIDI 27	Chorus FB
		MIDI 28	Chorus Rate
		MIDI 29	"ChorusDelay I/O (sys), InEQ: Treb boost"
		MPress	Vibrato
		ControlD	Amp cut
174	Cymbal Singers	MWheel	Vibrato
		Data	Layer 3 volume (ride cymbal)
		MIDI 22	"BandPass Width, HiPass Res"
		MIDI 23	Pan LFO adj
		MIDI 24	InEQ: Treb cut
		MIDI 25	(Aux) LaserVrb Level
		MIDI 26	LaserVrb contour
		MIDI 27	Pitch LFO Rate
		MIDI 28	Flange FB
		MIDI 29	Toggle: Pitcher + PitcherFlange
		MPress	"Vibrato, BandPass Freq"
		KeyNum	EnvCtl: Att+Dec
		GKeyNum	Pitcher Pitch+Weights
		PWheel	BandPass Freq
175	BriteBells^Glock	MWheel	Vibrato
		Data	Layer Toggle: BriteBells + Glock
		MIDI 22	Layer Levels ^ EnvCtl: Att
		MIDI 23	"Pitch adj, EnvCtl: Dec"
		MIDI 24	InEQ: Bass ^ EnvCtl: Imp
		MIDI 25	"(Aux) Room Level, (FX1) Hall Wet/Dry cut"
		MIDI 26	Room Decay Time
		MIDI 27	"Chorus Wet/Dry, Echo Wet/Dry"
		MIDI 28	"Chorus FB, Echo FB"
		MIDI 29	"Toggle: Chorus in/out, Hall + Echo out"
		MPress	Vibrato
176	Crystaline^RX7	MWheel	"Shaper ctl, Vibrato ^ Pan adj"
		Data	Toggle: Crystaline ^ RX7
		MIDI 22	"ShapeMod osc Pitch, Shape amt ^ LoPass Freq, Pitch adj"
		MIDI 23	"LoPass Res, EnvCtl: Att"
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Room Level
		MIDI 26	Room Decay Time+HFDamp
		MIDI 27	"Chorus Wet/Dry, Echo Wet/Dry"
		MIDI 28	"Chorus FB, Echo FB"
		MIDI 29	Toggle: Chorus + Echo
		MIDI 70	Layer AltCtl
177	Wave Power 2	MWheel	"Vibrato, In Pan adj"
		Data	"Layer enable, EnvCtl: Dec+Rel"
		MIDI 22	"Hall Wet/Dry, Layer Delay"
		MIDI 23	Hall Time
		MIDI 24	LaserVrb contour
		MIDI 25	"InEQ: Bass, Laser spacing"
		MIDI 26	"InEQ: Treb, Laser envelope "
		MIDI 27	Laser Level+FB+Pitch
		MIDI 28	LaserDelay Time
		MIDI 29	Pitcher Level
		MPress	Vibrato Layer
178	Trinibell	MWheel	Pitch ASR ctl
		Data	Hi+LoPass Freq
		MIDI 22	"Hi+LoPass Freq+Res, Pitch LFO"
		MIDI 23	"Enc Ctl - Att, Dist Drive adj"
		MIDI 24	EnvCtl: Rel
		MIDI 25	Tremolo Depth
		MIDI 26	Flange Wet/Dry
		MIDI 27	Chorus Mix
		MIDI 28	Delay FB (sys)
		MIDI 29	Flange FB boost
		MPress	"Dist Drive, Hi+LoPass Freq"

ID	Name		
179	Frozen^ Pizzibell	MWheel	Tremolo
		Data	Toggle: Frozen ^ Pizzibell
		MIDI 22	Hi+LoPass Freq ^ EnvCtl: Imp
		MIDI 23	"LoPass Res, Env Ctl - Att"
		MIDI 24	"LoPass Freq+Res, HiPass Freq ^ EnvCtl: Rel"
		MIDI 25	Delay Mix (sys)
		MIDI 26	Delay Time
		MIDI 27	Chorus Mix
		MIDI 28	"(Aux) Wet/Dry cut, InEQ: Bass"
		MIDI 29	Toggle: Flange I/O
180	Dencity	MWheel	Pitch LFO adj
		Data	"LoPass Freq, ChorDelay Wet/Dry"
		MIDI 22	LoPass LFO
		MIDI 23	"Pan ctl, EnvCtl: Att"
		MIDI 24	"Pitch ctl, EnvCtl: Rel"
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time
		MIDI 27	Delay Time (sys)
		MIDI 28	Delay Mix + FB
		MIDI 29	"Toggle: Chorus I/O, Delay Level"
181	Synth Bell 1^2	MWheel	"Vibrato, Pan adj, LoPass Res"
		Data	"Toggle: Synth Bells 1 + 2, AltCtl adj"
		MIDI 22	"LoPass Res, BandPass Width, EnvCtl: Rel"
		MIDI 23	Pan adj
		MIDI 24	Pitch LFO adj
		MIDI 25	"(Aux) Hall Level, (fx1) Chapel Wet/Dry"
		MIDI 26	"Hall HFDamp+Time, Chapel Time"
		MIDI 27	"Chapel preDelay, SRS center Freq adj"
		MIDI 28	"Chapel EarlyRef+Late Levels, SRS EQ adj"
		MIDI 29	Toggle: Chapel + SRS
		BKeyNu	EnvCtl: Att+Dec+Rel
		MPress	Vibrato
182	DynPercMix^Gator	MWheel	"LoPass Freq+Res, Layer EnvCtl: Rel+Att"
		Data	Toggle: DynPercMix + Gator
		MIDI 22	"Filt Freq+Res+Width, Pitch, EQ, EnvCtl: Att ^ Lo+AllPass Freq"
		MIDI 23	"LoPass Freq, Pitch adj ^ ParaMid+Treb Freq"
		MIDI 24	"BandPass Width, Layer Level + Xfade"
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall HFDamp + Decay Time
		MIDI 27	"Flange Wet/Dry, MDelay Wet/Dry"
		MIDI 28	"Flange FB, MDelay FB"
		MIDI 29	Toggle: Flange + TubeAmpDelayChorus
		ChanS	EnvCtl: Rel
		AttVel	EnvCtl: Dec

ID	Name		
183	Workingman Space	MWheel	slight Vibrato
		Data	"LoPass Freq, Pitch adj"
		MIDI 22	"HiPass Freq, HFstim, BandPass Width"
		MIDI 23	Pipe Pitch bend
		MIDI 24	Pipe Pitch adj
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time
		MIDI 27	Hall HFDamp
		MIDI 28	Hall Wet/Dry cut
		MIDI 29	Toggle: ChorusVrb + St Image
		MPress	slight Vibrato
184	OronicoKno^Shift	MWheel	Vibrato
		Data	Toggle: OronicoKno + Shift
		MIDI 22	"HFstim adj, Pan adj"
		MIDI 23	"InEQ: Bass, Layer Xfade"
		MIDI 24	"InEQ: Treb, Pan adj, EnvCtl: Rel"
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Decay Time+PreDelay
		MIDI 27	Delay Mix (sys)
		MIDI 28	Chorus Delay Time
		MIDI 29	Chorus Depth adj
		MPress	Vibrato
		AttVel	AltCtl
185	Ethereal Dawning	MWheel	BandPass Freq+Width sweep
		Data	Layer enable
		MIDI 20	BandPass Width - Layer 1
		MIDI 22	BandPass Freq + Width - Layer 2
		MIDI 23	BandPass Width - Layer 3
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Decay Time
		MIDI 27	Flange Wet/Dry
		MIDI 28	Flange FB
		MIDI 29	"Toggle: Flange + CDR, InEQ: Bass"
		MPress	BandPass Freq
186	RAVReligion^Prey	MWheel	"Vibrato, Para EQ adj"
		Data	Toggle: RAVReligion + Prey
		MIDI 22	EnvCtl: Att ^ Shapemod Pitch
		MIDI 23	"InEQ: Bass, Shape amt, EnvCtl: Rel ^ LoPass Freq, EnvCtl: Dec+Rel"
		MIDI 24	"InEQ: Treb, LoPass Freq, EnvCtl: Rel + Dec"
		MIDI 25	Toggle: (Aux) Hall + (fx1) Room cut
		MIDI 26	"Hall Decay Time, Room Time"
		MIDI 27	Phaser center Freq
		MIDI 28	Phaser FLFO Depth
		MIDI 29	Toggle: Room + Phaser
		MPress	Vibrato

ID	Name		
187	Voyage	MWheel	"Vibrato, LoPass Freq"
		Data	Layer enable
		MIDI 22	"Pitch LFO Rate, LoPass Freq+Res, EnvCtl: Imp+Att+Rel"
		MIDI 23	EnvCtl: Dec
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	Hall Time+HFDamp
		MIDI 27	Chorus Mix
		MIDI 28	Chorus FB
		MIDI 29	"Chorus Rate, St Image L/R Delay"
		MPress	"Vibrato, LoPass Freq"
		MIDI 70	AltCtl
188	WispSingrs^Glass	MWheel	"Vibrato, LoPass Res"
		Data	Toggle: WispSingrs + Glass
		MIDI 22	"LoPass Freq+Res, HiPass Freq"
		MIDI 23	"LoPass Freq, HiPass Res+Freq, Layer Levels"
		MIDI 24	EnvCtl: Att+Rel
		MIDI 25	(Aux) Hall + (fx1) Hall Wet/Dry
		MIDI 26	Hall Times+HFDamp
		MIDI 27	Chorus Wet/Dry
		MIDI 28	Delay Wet/Dry (sys)
		MIDI 29	Toggle: Hall + CDR
		MPress	Vibrato
189	Luscious	MWheel	Vibrato
		Data	"Panner LFO Rate, Layer Delay, Layer Xfade"
		MIDI 22	EnvCtl: Imp+Att
		MIDI 23	"InEQ: Bass, EnvCtl: Dec"
		MIDI 24	"InEQ: Treb, EnvCtl: Rel"
		MIDI 25	(Aux) Hall Time+PreDelay+HFDamp
		MIDI 26	Flange Mix
		MIDI 27	Flange Rate
		MIDI 28	Flange FB
		MIDI 29	Hall PreDelay adj
		MPress	Vibrato
		Tempo	LoPass Freq
190	LightMist^Padify	MWheel	Vibrato
		Data	Toggle: LightMist + Padify
		MIDI 22	"Pitch adj, LoPass Freq"
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level
		MIDI 26	Chorus Delay Time
		MIDI 27	Chorus Delay Depth
		MIDI 28	Delay Mix (sys)
		MIDI 29	Hall Time+PreDelay adj
		MPress	Vibrato
191	VortexRev^Launch	MWheel	Vibrato
		Data	Toggle: VortexRev + Launch
		MIDI 22	"HiPass Freqs+Width, EQ adj"
		MIDI 23	"InEQ: Bass, Layer Xfade"
		MIDI 24	"InEQ: Treb, EnvCtl: Att+Rel"
		MIDI 25	(Aux) Hall Time
		MIDI 26	Hall PreDelay
		MIDI 27	Chorus Depth+Delay
		MIDI 28	Delay Mix+FB
		MIDI 29	Hall HFDamp
		MPress	Vibrato

ID	Name		
192	Hello^A No Way	MWheel	LoPass Freq
		Data	Toggle: Hello + A No Way
		MIDI 22	"LoPass Freq LFO, Pitch mod"
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level+Time
		MIDI 26	Hall build+PreDelay
		MIDI 27	Chorus FB
		MIDI 28	Chorus Delay
		MIDI 29	Delay Mix
		MPress	Pitch mod
193	Environments	MWheel	"""hi bird"" LFO Rate, Panner adj"
		Data	"""lo bird"" LFO Rate"
		MIDI 22	"ParaEQ Freq, shaper amt"
		MIDI 23	"Pitch adj, LoPass Freq, BandPass Freq+Width"
		MIDI 24	"HiPass Freq, Pitch (sine)"
		MIDI 25	"Chorus Level, rvb Level, CDR Wet/Dry"
		MIDI 26	(fx2) Chorus Wet/Dry
		MIDI 27	Phaser Wet/Dry
		MIDI 28	CDR Wet/Dry
		MIDI 29	Chorus Rate
		MPress	InEQ: Bass
		MIDI 70	AltCtl
194	Gremlin Groupies	MWheel	"Layer Pitch, LoPass Freq+Res, Wrap adj"
		Data	"Layer Pitch, LoPass LFO adj"
		MIDI 22	"Layer Pitch, Pitch (Sine) adj"
		MIDI 23	Layer Pitch adj
		MIDI 24	"Layer Pitch, Wrap adj"
		MIDI 25	(Aux) Hall Level
		MIDI 26	"Pitcher Wet/Dry, LsrDelay Time+Wet/Dry"
		MIDI 27	"Pitcher wts pair, Lsr Spacing"
		MIDI 28	"Pitcher wts odd, Lsr Contour"
		MIDI 29	Toggle: Pitcher + LaserDelay
195	Lost In Space	MWheel	"LoPass Freq adj, Pitch adj"
		Data	HiPass Freq adj
		MIDI 22	LoPass + HiPass Freq adj
		MIDI 23	"Layer Pitch, EnvCtl- atk"
		MIDI 24	EnvCtl- Rel
		MIDI 25	Delay Mix
		MIDI 26	Delay Time
		MIDI 27	Ch FB
		MIDI 28	Ch Depth
		MIDI 29	Toggle: (Aux) Plate I/O
		MPress	
196	Lunar Wind	MWheel	LoPass Freq+Res
		Data	Pitch adj
		MIDI 22	"LoPass Res, Pan adj"
		MIDI 23	Panner sweep
		MIDI 25	(Aux) Room Level
		MIDI 26	Pitcher Wet/Dry
		MIDI 27	Flange Mix (sys)
		MIDI 28	Pitcher Pitch
		MIDI 29	Toggle: Pitcher I/O
		MPress	"LoPass Freq, Pan LFO"
		ChanS	EnvCtl: Rel
		Breath	LoPass adj

ID	Name		
197	Noise Toys	MWheel	"Pitch LFO, Shaper amt"
		Data	"Pitch (Sine+) adj, BandPass Freq, Dist amt"
		MIDI 22	"Pitch adj, Shaper LFO, HiPass Freq"
		MIDI 23	"LoPass + HiPass Freq, EnvCtl: Att"
		MIDI 24	EnvCtl: Rel
		MIDI 25	(Aux) Hall Level
		MIDI 26	"LrsDelay Wet/Dry, Pitch Wet/ Dry"
		MIDI 27	"LsrDelay contour, Pitch pair weights"
		MIDI 28	Pitch odd weights
		MIDI 29	Toggle: Laser + Pitch
		MPress	"Vibrato, Pitch LFO adj"
		PWheel	Shaper adj
		Tempo	LsrDelay Delay coarse + spacing
770	Concert Piano 2	MIDI 25	(Aux) Hall Level+Time
771	Studio Grand	MIDI 25	(Aux) Hall Level+Time
		MIDI 29	Soundboard Rvb Enable
			(Soft Pedal is Active)
772	Soft Piano	MIDI 25	(Aux) Hall Level+Time
		Sustain	Soundboard Wet/Dry Increase
773	MonoStudioGrand	Data	ParaTreb Gain
		MIDI 25	(Aux) Hall Level+Time
			(Soft Pedal is Active)
774	RandomPan Grand	MIDI 25	(Aux) Hall Level+Time
775	Funky Piano	MWheel	ParaEQ LFO Depth
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Room Level+Time
		MIDI 26	Flange Wet/Dry
		MIDI 27	Flange FB
		MIDI 28	Flange XCouple
		MIDI 29	Flange LFO Tempo
		MPress	ParaEQ Depth
776	Piano Chase	MWeel	Vibrato (Strings)
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Plate Level+Time
		MIDI 26	Flange Wet/Dry
		MIDI 27	"Flange FB, (Aux) Decay Time"
		MIDI 28	Flange LFO Tempo
		MIDI 29	Flange XCouple
		MPress	Vibrato (Strings)
		Sost Ped	Disables Strings
777	Grand+Elec 1	MWheel	Layer Balance
		MIDI 25	"(Aux) Hall Level, Room Wet/ Dry"
778	Grand+Elec 2	MWheel	E Pno Vibrato + ParaTreb
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level
		MIDI 26	Chorus Wet/Dry
		MIDI 27	Chorus FB
		MIDI 28	Chorus XCouple
		MIDI 29	(Aux) Early Ref Level
		Soft Ped	Softens Elec Piano
779	Detuned Piano	MWheel	"Vibrato, Tremolo"
		Data	Detune
		MIDI 25	(Aux) Hall Level
		MIDI 26	(Aux) Hall Time
		MIDI 27	Room Wet/Dry

ID	Name		
780	Ballad Piano	MWheel	Tremolo
		Data	Layer Enable/Disable
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treb
		MIDI 25	(Aux) Hall Level
		MIDI 26	(Aux) Hall Time
		MIDI 29	Soundboard I/O
781	Piano&SpaceyPad	MWheel	Pad Balance
		MIDI 22	Notch Filt Freq+Width
		MIDI 23	Panner
		MIDI 24	EnvCtl: Release (Pad)
		MIDI 25	(Aux) Hall Level
		MIDI 26	(Aux) Hall Time
		MIDI 29	Toggle: Soundboard I/O
782	Lush Piano Pad	MWheel	"Vibrato, Pan adj"
		Data	"Layer Enable, Delay "
		MIDI 22	"EnvCtl: Att+Dec, Rel (Layers1+2)"
		MIDI 23	"EnvCtl: Dec (Layers 3+4), InEQ: Bass"
		MIDI 24	"EnvCtl: Rel (Layers 3+4), InEQ: Treb"
		MIDI 25	(Aux) Hall Wet/ Dry+PreDelay+HFDamp
		MIDI 26	Flange Mix
		MIDI 27	Flange Rate+Xcurs
		MIDI 28	Flange FB
		MPress	"Vibrato, Layer Balance"
783	Rotating Piano	MWheel	Leslie Enable
		MIDI 25	(Aux) Plate Level
		MIDI 26	Plate Time
		MIDI 27	Hi/Low Tremolo adj
		MIDI 28	(Aux) HFDamp
		MIDI 29	Toggle: VibratoChorus I/O
		Soft Ped	Leslie Enable
784	MajesticPianoism	MWheel	Vibrato/Tremolo
		Data	Disable Strings
		MIDI 22	LoPass Freq+Res (Strings)
		MIDI 25	(Aux) Hall Level
		MIDI 26	(Aux) Rvb Time
		MIDI 27	(insert) Room Wet/Dry
		AttVel	Enables Dull Saw Wave
785	Sonar Piano	MWheel	Vibrato (clave)
		MIDI 25	(Aux) Hall Level
		PWheel	Octave Shift
786	Pop Grand	Data	Enhc Low Mix
		MIDI 22	Enhc Mid Mix
		MIDI 23	Enhc Hi Mix
		MIDI 25	(Aux) FDR Level
		MIDI 26	Flange Wet/Dry
		MIDI 27	Flange Tempo
		MIDI 29	(Aux) Delay Mix
787	Grandsichord	MWheel	Vibrato
		Data	"Slight detune, EnvCtl: Att+Rel"
		MIDI 22	Hi Freq Stimulator Cut
		MIDI 25	(Aux) FDR Level
		MIDI 26	Flange Wet/Dry
		MIDI 27	Flange Tempo
		MIDI 28	Enhc Lo/Mid Drive
		MIDI 29	(Aux) Delay Mix

Monaural Piano Programs

Most of the piano programs are set to play in stereo, though **773 MonoStudioGrand**, as well as a number of programs on the accessory disk are designed for mono use. If the pianos are to be played through a mono sound system, the best results will come from these mono programs, not the stereo programs mixed to mono.

Stretch Tuning

Unless otherwise noted, piano programs are "stretch" tuned, like an acoustic piano. Since the higher harmonics of a stretched string tend to be sharper than those of the real harmonic series, stretch tuning ensures that the piano remains in tune with itself harmonically. Stretch tuning is sometimes referred to as "solo" or "beat" tuning.

Keymaps with **440** as part of their name—like **776 Mono Piano 440**—offer straight (non-stretch) tuning, where the fundamental of each note is tuned to A440. Programs that use these keymaps (for example **780 Ballad Piano**) will mix better with other acoustic and electronic instruments. This type of tuning, therefore, is sometimes known as "ensemble" tuning.

Appendix D
Contemporary ROM Block Objects

In This Appendix

The objects listed in this Appendix are current with operating system version 1.01. Your K2600 probably has version 1.01 objects installed. Here's how you can check the version of the objects you have installed:

1. Press the **Master** mode button to enter Master mode.
2. Select the Intonation parameter.
3. Change its value to **19**. You should see something like this: **19 Obj C1.00**. This "intonation table" is actually the version number of the K2600's basic object file (hence the **C**). If you scroll higher in the list, you may see other version numbers, depending on the ROM block options you have.

If your instrument doesn't have version 1.01 objects, you can get them from our website:

http://www.youngchang.com/kurzweil/html/downloads.html

Programs

Pianos	
794	Water Piano
795	StPno & OrchPad
796	Grand & Pad
797	Pop Grand Stack
798	Prepared Piano
799	Tack Piano Stack
Ethnic / World Instruments	
800	Jungle Jam
801	Mbira Stack
802	Ritual Metals
803	Prepared Mbira
804	Balinesque
805	Ambient Bells
806	World Jam 1
807	World Jam 2
808	India Jam
809	Slo Wood Flute
810	Hybrid Pan Flute
811	Chiff Brass Lead
812	Bell Players
813	Prs Koto
814	Medicine Man
815	Mbira
816	Kotobira
817	Cartoon Perc
818	CowGogiBell
819	Perc Pan Lead
820	Trippy Organ
821	Koto Followers
822	Hybrid Horn
Keyboards	
823	Dyno EP Lead
824	ParaKoto
825	Super Clav
826	StrataClav
827	Touch Clav
828	Bad Klav
829	Rad Rotor
830	B-2001
831	Perc Organ
832	Drawbar Organ CS
Brass and Reeds	
833	Bebop Alto Sax
834	Soft Alto Sax
835	Soprano Sax
836	Low Soft Sax
837	Air Reeds CS
838	Jazz Muted Trp
839	Jazz Lab Band
840	Harmon Section
841	Sfz Cres Brass
842	Neo Stabs
843	Gtr Jazz Band
844	Full Rock Band

Drum Kits	
845	World Rave Kit
846	Punch Gate Kit
847	Shadow Kit
848	Fat Traps
849	Generator Kit
850	Shudder Kit
851	Crowd Stomper
852	Econo Kit
853	EDrum Kit 1
854	EDrum Kit 2
Loops	
855	Dog Chases Tail
856	Saw Loop Factory
Basses	
857	Two Live Bass
858	Dual/Tri Bass
859	Clav-o-Bass
860	Chirp Bass
861	DigiBass
862	Mono Synth Bass
863	Touch MiniBass
864	Ostinato Bass
865	House Bass
866	Dubb Bass
Guitars	
867	Straight Strat
868	Chorus Gtr
869	Strataguitar
870	Elect 12 String
871	Dyn Jazz Guitar
872	Pedal Steel
873	Strummer DistGtr
874	Rock Axe
875	Hammeron
876	Rock Axe mono
Synths	
877	Attack Stack
878	Skinny Lead
879	Q Sweep SynClav
880	Anna Mini
881	Ballad Stack
882	Big Stack
883	BrazKnuckles
884	Hybrid Breath
885	Hybrid Stack
886	Eye Saw
887	Mello Hyb Brass
888	Sizzl E Pno
889	My JayDee
890	Slo SynthOrch
891	SpaceStation
892	Glass Web
893	Circus Music
894	Mandala
895	Slow Strat
896	Fluid Koto
897	Koreana Pad
898	Tangerine
899	Planet 9

Setups

800	HyperGroov<-C4->
801	PianoPad w/Percs
802	Slo Held Arper
803	Don'tGetFooled
804	Touch Game
805	BeatBoy E1
806	ZawiClav Split
807	Dyn Piano Pad
808	Pulsar Stack
809	Mt Chicorora C2
810	Hold Low 3sec Rb
811	Mettlorfus Pad
812	Black Keys xtra
813	Jungle Jammer
814	Huge Rock Band
815	Rock Ballad
816	Jazz Setup
817	Two Touchers
818	Frontier prs
819	Eclectric Grand
820	Bad Trip FtSw/MW
821	WhirliToys
822	PluckSynths Perc
823	SusPed RhythmJam
824	Ballad Piano Pad
825	Big AnaLoveVibe
826	ShockBreaks PSw1
827	Four Pluckers
828	WaterPiano Pad
829	Padded Room
830	AtmosPolySphere
831	Breath Pad
832	Trippy Jam
833	MeditationGuits
834	Cool Down Funk
835	Tek`Groov C5->
836	Big Fat Split
837	The Pump C2
838	Ana Basses
839	Multi Followers
840	Plucksynths
841	10 Leagues Under
842	Gremlin Arps
843	Broken Toys
844	Two Synth
845	Machine Shop
846	Farawaway Place
847	BehindEnemyLines
848	Tunnel Visionprs
849	Seismic Trance
850	Medal

QA Banks

800	Bands
801	Grooves
802	World
803	Pop
804	More Keys
805	More Analog
806	Leads
807	Trio Parts
808	Techno
809	Texture

Keymaps

800	Hybrid Pan
801	Glass Rim Tone
802	Synth Vox
803	Orch Pad
804	Koreana
805	Heaven Bells
806	MIDI Stack
807	Synth Brass
808	DigiBass
809	AnaBass
810	Mini Saw
811	EBass Pick
812	EBass Slap
813	Clean Elec Gtr
814	Distorted Guitar
815	Dist Harmonics
816	Clav
817	Tone Wheel Organ
818	Muted Trumpet
819	Soft Alto Sax
820	Koto
821	Mbira
822	Tabla Ta
823	Tabla Tin
824	Tabla Dhin
825	Tabla/Bayan Dha
826	Bayan
827	Ghatam Bass Tone
828	Small Ghatam
829	Ghatam Shell
830	Ghatam Slap
831	Dumbek Open Tone
832	Dumbek Brt Tone
833	Dumbek Tek
834	Dumbek Snap
835	Dumbek Dry Dum
836	Djembe Tone
837	Djembe Cl Slap
838	Djembe Open Slap
839	Djembe Finger
840	Djembe w/ Stick
841	Muzhar
842	Talking Drum Lo
843	Talking Drum Hi
844	Luna Drum Dry
845	Luna Drum Hi
846	Log Drum Lo
847	Log Drum Hi
848	Shakers/Tamborim
849	Gankogui Bell Lo
850	Gankogui Bell Hi
851	Tibetan Cymbal
852	Tibetan Bowl
853	Indo Bowl Gong
854	Prev Ethnic Perc
855	Cartoon Perc
856	Prev EDrum Map
857	Toms Map
858	ProcKick/Snr Map
859	EDrum Kit 1
860	EDrum Kit 2
861	1 Lyr Proc Kit
862	Industry Perc
863	Tuned Loops
870	PreparedMbira L1
871	PreparedMbira L2
872	World Jam 1 L1
873	World Jam 1 L2
874	World Jam 1 L3
875	India Jam L1
876	India Jam L2
877	World Jam 2 L1
878	World Jam 2 L2
879	World Jam 2 L3
880	World Jam 2 L4
881	World Jam 2 L5
882	World Jam 2 L6
883	World Jam 2 L7
884	World Jam 2 L8
885	CowGogiBell L1
886	Dual Log Drum
887	Jungle ProcDrms
888	JungleBrushTip1
889	JungleBrushTip2
890	Jungle Birds
891	Jungle Ghtm rel
892	Jungle Tabla
893	Jungle Dumbek
894	Jungle ProcDrms2
895	Jungle GhtmStrgt
896	Syn Bass Pick
897	ARP SAW
898	ARP PW30%
899	OB PW25%

Samples

800	Hybrid Pan
801	Glass Rim Tone
802	Synth Vox
803	Orch Pad
804	Koreana
805	Heaven Bells
806	MIDI Stack
807	Synth Brass
808	DigiBass
809	AnaBass
810	Mini Saw
811	EBass Pick
812	EBass Slap
813	Clean Elec Gtr
814	Distorted Guitar
815	Dist Harmonics
816	Clav
817	Tone Wheel Organ
818	Muted Trumpet
819	Soft Alto Sax
820	Koto
821	Mbira
822	Tabla Ta
823	Tabla Tin
824	Tabla Dhin
825	Tabla/Bayan Dha
826	Bayan
827	Ghatam Bass Tone
828	Small Ghatam
829	Ghatam Shell
830	Ghatam Slap
831	Dumbek Open Tone
832	Dumbek Brt Tone
833	Dumbek Tek
834	Dumbek Snap
835	Dumbek Dry Dum
836	Djembe Tone
837	Djembe Cl Slap
838	Djembe Open Slap
839	Djembe Finger
840	Djembe w/ Stick
841	Muzhar
842	Talking Drum Lo
843	Talking Drum Hi
844	Luna Drum Dry
845	Luna Drum Hi
846	Log Drum Lo
847	Log Drum Hi
848	Shakers/Tamborim
849	Gankogui Bell Lo
850	Gankogui Bell Hi
851	Tibetan Cymbal
852	Tibetan Bowl
853	Indo Bowl Gong
854	EDrum1 Kick
855	EDrum1 Snare
856	EDrum1 Rim
857	EDrum1 Hi Tom
858	EDrum1 Crash
859	EDrum1 Cowbell
860	EDrum1 Clave
861	EDrum1 Shaker
862	EDrum2 Kick1
863	EDrum2 Kick2
864	EDrum2 Kick3
865	EDrum2 Snare1
866	EDrum2 Snare2
867	EDrum2 Snare3
868	EDrum2 HH Open
869	EDrum2 HH Close
870	EDrum2 Clap
871	EDrum2 Conga
872	Hi Proc Tom
873	Hi Mid Proc Tom
874	Lo Mid Proc Tom
875	Lo Proc Tom
876	Syn Toms
877	Proc Kicks
878	Proc Snares
879	Rvs Proc Kicks
880	Rvs Proc Snares
881	Bayan Mute
882	Alt Muzhar Rim
883	Alt Tabla Ta
884	Alt Maracas
885	Alt Shakere
886	Syn Bass Pick
887	Alt Log Drum Lo
888	Alt Tibetan Cym
891	Dumbek Mute Slap
896	ROM Loops
897	ARP SAW
898	ARP PW30%
899	OB PW25%

Program Control Assignments

The preset programs in the K2600 Contemporary ROM option are organized by category. You can either use them as they are or as a good starting point for your own work. There are many ways to put expressivity and variety in a single program by assigning controllers to the various DSP functions in its layers. This list describes how each of the preset programs can be modulated or altered by various controllers. Only those control assignments that may not be immediately evident are listed. Control assignments like attack velocity and keynumber apply to most programs.

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
Pianos					
794	Water Piano	Vibrato	Wet/Dry mix	Vibrato	
795	StPno & OrchPad	Pad balance			
796	Grand & Pad	Pad balance	Bell release envelope		
797	Pop Grand Stack	Bell fade	Wet/Dry mix	Vibrato	
798	Prepared Piano	Alt switch - mbira	Wet/Dry mix		
799	Tack Piano Stack	Bell fade, Wet/Dry mix	Pitch env - mbira		
Ethnic / World Instruments					
800	Jungle Jam	This program uses the mirror image drum mapping, symmetrical around D4. Identical or similar drum articulations are found at equal distances above and below D4, with extras outside the center region. Mod wheel disables layered “chirps" and fades rain stick on A0. Data slider enables "screamers" on G5-C6.			
801	Mbira Stack	Vibrato			
802	Ritual Metals	Vibrato		Vibrato	
803	Prepared Mbira		Pitch change		
804	Balinesque	Pan flute fade			
805	Ambient Bells	Vibrato		Vibrato	
806	World Jam 1		Pitch change		Mirror image drum mapping
807	World Jam 2		Pitch change	Layer pitch	Mirror image drum mapping
808	India Jam	Tablas appear at center with the mirror-image mapping, tuned to C. Pressure controls pitch for the bayan and RH lead sound. LH drone may be played as broken chord C2,G2,C3,G3 and held with sustain or sostenuto. Mod Wheel fades the drone. Data Slider controls Wet/Dry mix.			
809	Slo Wood Flute	Less tremolo		Filter ctl	
810	Hybrid Pan Flute	Tremolo		Tremolo	
811	Chiff Brass Lead	Vibrato, Swell	Unison layers	Vibrato, Filter	
812	Bell Players	Muzhar fade	Tibetan cym env ctl		
813	Prs Koto			Pitch mod	
814	Medicine Man				
815	Mbira	Release ctl	Tremolo		
816	Kotobira	Mbira balance			
817	Cartoon Perc		Wet/Dry mix		
818	CowGogiBell	Alt start	Layer select		
819	Perc Pan Lead	Vibrato			
820	Trippy Organ	Vibrato		Vibrato	
821	Koto Followers	Vibrato		Vibrato	
822	Hybrid Horn	Balance (bell)		Timbre ctl, Vibrato	

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
Keyboards					
823	Dyno EP Lead	Tremolo, Env ctl			
824	ParaKoto	Pad tremolo			
825	Super Clav	Phase clav enable	Disable release	Filter rate	
826	StrataClav	Vibrato		Vibrato	
827	Touch Clav	EQ, Vibrato	Disables release	Filter control	
828	Bad Klav				
829	Rad Rotor	Rotary speaker			
830	B-2001	Rotary speaker	Perc balance	Rotary speaker	
831	Perc Organ	Rotary speaker	Perc balance	Rotary speaker	
832	Drawbar Organ CS	Rotary speaker	Filter ctl		
Brass and Reeds					
833	Bebop Alto Sax	Attack ctl		Vibrato	
834	Soft Alto Sax			Vibrato, Swell	
835	Soprano Sax	Vibrato, Swell		Vibrato, Swell	
836	Low Soft Sax			Vibrato	
837	Air Reeds CS	Vibrato	Harmonica enable	Harmonica vibrato	
838	Jazz Muted Trp				
839	Jazz Lab Band			Vibrato, Swell	
840	Harmon Section	Vibrato		Vibrato, Swell	
841	Sfz Cres Brass	Vibrato	Wet/Dry mix	Vibrato, Swell	
842	Neo Stabs	Vibrato		Vibrato, Filter ctl	
843	Gtr Jazz Band	LH bass is layered with ride for walking rhythm section. LH hard strikes trigger kick/snare. Data slider switches RH from guitar to horn section; SostPed holds horns and adds bright tenor.			
844	Full Rock Band	LH bass is layered with kick/snare for driving rhythm section. At *ff*, crash cymbal is triggered. Mod wheel and pressure enable rotary speaker for RH organ. Data slider switches LH to walking rhythm section, and RH to guitar solo.			
Drum Kits					
845	World Rave Kit	Disable chirps	Wet/Dry mix, Disable claps (G6-G#6)		
846	Punch Gate Kit		Wet/Dry mix		
847	Shadow Kit	Flanging (A#3-B3)	Wet/Dry mix		
848	Fat Traps	Filter (C2-A#2)	Wet/Dry mix		
849	Generator Kit	Disable claps (G3-G#3)	Wet/Dry mix		
850	Shudder Kit		Wet/Dry mix		
851	Crowd Stomper		Wet/Dry mix		
852	Econo Kit	Gate time (G3-C#4)	Wet/Dry mix		
853	EDrum Kit 1	Gate time (B2-D#3, G3-C#4), Pitch (D6)	Wet/Dry mix	Pitch (D6)	Sust ped chokes cymbal (F#5)
854	EDrum Kit 2	Filter ctl (A#1-C2, F#6-C7)	Wet/Dry mix		
Loops					
855	Dog Chases Tail	Various loop effects	Tempo (pitch)		Loops below E4 are tuned to play together, as are loops above E4.
856	Saw Loop Factory	Layer balance	Tempo (pitch)		

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
Basses					
857	Two Live Bass	Vibrato	Layer select	Vibrato	
858	Dual/Tri Bass	Vibrato	Ghost note enable	Vibrato	
859	Clav-o-Bass	Vibrato	Wet/Dry mix	Vibrato	
860	ChirpBass	Vibrato	Wet/Dry mix	Vibrato	
861	DigiBass				
862	Mono Synth Bass		Filter		Pitch bend goes +2/-12ST
863	Touch MiniBass	Vibrato		Vibrato, Swell	
864	Ostinato Bass		EQ		
865	House Bass	Vibrato	Release ctl	Vibrato	
866	Dubb Bass	Vibrato	Release ctl	Vibrato	
Guitars					
867	Straight Strat	Tremolo	EQ		
868	Chorus Gtr		Wet/Dry mix	Detune	
869	Strataguitar	Alt start			
870	Elect 12 String	Detune	Wet/Dry mix, EQ	Vibrato	
871	Dyn Jazz Guitar		Wet/Dry mix		PBend gives fretboard slide
872	Pedal Steel	Vibrato		Vibrato	
873	Strummer DistGtr	Vibrato		Vibrato	
874	Rock Axe	Alt start	EQ	Feedback	
875	Hammeron	Timbre ctl		Timbre ctl	
876	Rock Axe Mono	Alt start	EQ, Delay	Feedback	
Synth Timbres					
877	Attack Stack	Vibrato	Wet/Dry mix	Vibrato	
878	SkinnyLead	Vibrato	Overdrive enable	Vibrato, Filter	
879	Q Sweep SynClav	Vibrato	Sweep rate ctl	Vibrato	
880	Anna Mini	Vibrato		Vibrato	
881	Ballad Stack	Swell		Swell	
882	Big Stack	Vibrato	Env ctl	Vibrato	
883	BrazKnuckles	Swell	EQ		
884	Hybrid Breath	Envelope ctl, EQ	Envelope ctl, Wet/Dry mix	Vibrato	
885	Hybrid Stack		Layer balance		
886	Eye Saw	Vibrato	Release ctl, Filter	Vibrato	
887	Mello Hyb Brass				
888	Sizzl E Pno	Pad balance			
889	My JayDee	Vibrato	Release ctl	Vibrato	
890	Slo SynthOrch	Filter effect			
891	SpaceStation	Vibrato	Envelope ctl	Vibrato	
892	Glass Web	EQ	Delay ctl		
893	Circus Music	Vibrato		Vibrato	

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
Pads					
894	Mandala	Filter ctl	Pitch change		
895	Slow Strat	Vibrato	Filter sweep enable	Vibrato	
896	Fluid Koto	Vibrato		Vibrato	
897	Koreana Pad	Tremolo	Filter, Wet/Dry mix		
898	Tangerine	Enable 5th	Envelope Ctl	Vibrato	
899	Planet 9				

Appendix E
Orchestral ROM Block Objects

In This Appendix

The objects listed in this Appendix are current with operating system version 1.01. Your K2600 probably has version 1.01 objects installed. Here's how you can check the version of the objects you have installed:

1. Press the **Master** mode button to enter Master mode.
2. Select the Intonation parameter.
3. Change its value to **20**. You should see something like this: **20 Obj O1.00**. This "intonation table" is actually the version number of the K2600's basic object file (hence the **O**). If you scroll higher in the list, you may see other version numbers, depending on the ROM block options you have.

If your instrument doesn't have version 1.01 objects, you can get them from our website:

http://www.youngchang.com/kurzweil/html/downloads.html

Programs

Pianos	
788	Piano Trio
789	Pno & Syn String
790	Fluid Grand
791	Haunted Piano
792	Xylopiano
Orchestras	
793	Grand,Harp&Lead
900	TotalCntrl Orch1
901	TotalCntrl Orch2
902	BaroqueOrchestra
903	Oboe&Flute w/Str
904	Horn&Flute w/Str
905	Trp&Horns w/Str
Winds	
906	Piccolo
907	Orchestral Flute
908	Solo Flute
909	Orchestral Oboe
910	Solo Oboe
911	2nd Oboe
912	Orch EnglishHorn
913	Solo EnglishHorn
914	Orch Clarinet
915	Solo Clarinet
916	Orch Bassoon
917	Solo Bassoon
918	Woodwinds 1
919	Woodwinds 2
Brass	
920	Dynamic Trumpet
921	Copland Sft Trp
922	Orch Trumpet
923	Soft Trumpet
924	Strght Mute Trp
925	French Horn MW
926	Slow Horn
927	F Horn Con Sord
928	F Horns a2 MW
929	French Horn Sec1
930	French Horn Sec2
931	Solo Trombone
932	Tuba
933	Dyn Hi Brass
934	Dyn Lo Brass
935	Dyn Brass & Horn
936	Soaring Brass
937	MarcatoViolin MW
938	Solo Violin
939	2nd Violin
940	Orch Viola
941	Solo Viola
942	Slow Viola
Solo Strings	
943	Marcato Cello MW
944	Solo Cello
945	Slow Cello
946	Arco Dbl Bass
947	Slow Arco Bass
948	Brt Dbl Bass

String Sections	
949	Touch Strings
950	Fast Strings MW
951	Chamber Section
952	Sfz Strings MW
953	Sweet Strings
954	Baroque Strg Ens
955	Big String Ens
956	Bass String Sec
957	Pizzicato String
958	Wet Pizz
959	Arco & Pizz
Plucked Strings	
960	Classical Guitar
961	Virtuoso Guitar
962	Acoustic Bass
963	Snappy Jazz Bass
964	Dynamic Harp
965	Harp w/8ve CTL
966	Harp Arps
Keyboards	
967	Celesta
968	Pipes
969	Pedal Pipes 2
970	Church Bells
971	Glockenspiel
Percussion	
972	Xylophone
973	Chimes
974	Timpani/Chimes
975	Timpani
976	Timpani & Perc
977	Big Drum Corp
978	Orch Percussion1
979	Orch Percussion2
980	Jam Corp
981	Conga & Perc
982	Woody Jam Rack
983	Metal Garden
984	Hot Tamali Kit
985	Funk Kit
986	Magic Guitar
987	Glass Bow 2
988	Synth Orch
989	Nooage InstaHarp
990	AC Dream
991	Synth Dulcimer
992	Glistener
993	Afro Multi CTL
994	Tranquil Sleigh
995	Batman Strings
996	Ethnoo Lead
997	Orch Pad CTL
998	Choral Sleigh
999	Pad Nine

Setups

900	Deep Piano Rbn
901	Choir & Harp
902	Orchestrator
903	Piano Concerto
904	Xmas Carols
905	Sideline Perc
906	TonalGroov C5->
907	Exotic Grooves
908	Lunar Harp
909	Themes
910	Wet Piano
911	Enter the Jester
912	Tap the Jester
913	Hybrid Strings
914	Wonderous Spaces
915	Metal Orch Pad
916	Toon prs
917	Tranquil Sea
918	Sick Clock Jam
919	Orc Split
920	Baroque Brass
921	Unison Orchestra
922	Unison w/Pizz
923	Switch Orchestra
924	Pizz/Str/Winds
925	Harp Arps Cmaj
926	Desert Bloom E1
927	Exotic Charge
928	ET Comes Home
929	Fanfare Orch
930	Switch Orch 2
931	Orbiting Venus
932	Glass Dulcimer
933	Hybrid Reeds
934	Two Hand Pizz
935	Slo Str & Horn
936	Pianist Band
937	Prepared Pianos
938	FSW1 solo winds
939	Strings&Winds
940	Str Ens Solo MW
941	Pno&Vox&Pizz
942	Down Wind SmRbn
943	Guitar & Piano
944	Cirrus 9
945	Dry Plucks
946	String Collage
947	Esoterica
948	Poseidon
949	Stalkers
950	Diabolic Trickle

QA Banks

900	Piano Patch
901	Full Orch
902	Strings
903	Horns
904	Winds
905	Solo Orch
906	Perc Pit
907	Perc Ens
908	Moody
909	Exotic

Keymaps

900	Oboe
901	English Horn
902	Bassoon
903	Clarinet
904	Bassoon/Oboe
905	Bsn/EHrn/Oboe
906	Flute 2
907	Eng Horn/Oboe
910	Soft Trumpet
911	French Horn
912	French Hrn Sec
913	Tuba
914	Tuba/Horn
915	Tuba/Hrn Sec
916	Tuba/Sft Trmp
917	Trombet
918	Trumpbone
919	Trombne/SftTrmpt
920	Timpani
921	Snare Roll
922	Snare Hit
923	Orch Bass Drum
924	Orch Crash
925	Tam Tam
926	Triangle
927	Tambourine Roll
928	Tamb Hit
929	Sleigh Bells
930	Woodblock
931	Low Clave
932	Castanet Hit
933	Castanet Up
934	Dry Snares
935	Amb Snares
936	Bass Drums
937	Orch Perc Units
938	Orch Perc Full
939	Misc Percussion
940	2Hand Amb Kit
941	2Hand Dry Kit
942	2H Kit Unit1
943	2H Kit Unit2
944	Xylophone
945	Glockenspiel
946	Chimes
947	2Hand DrumCorp
948	Lite Metal
949	Woody Perc
950	Celeste
951	Plucked Harp
952	Harp Gliss
953	Nylon String Gtr
954	Nylon Str noA2
955	Nylon for dulc
957	Acoustic Bass
960	Pizz Strings
961	Full Kbd DblBass
962	Solo Violin
963	Solo Viola
964	Solo Cello
965	fast Solo Cello
966	Solo Double Bass
967	Bass/Cello
968	Bass/Cello/Vio
969	Cello/Vla/Cello
970	Cello/Vla/Vln
971	Ens Strings 2
972	Solo Section 1
973	Solo Section 2
978	Harparps 2
979	BassDrum/Timp
980	Organ Wave 8
981	Buzz Wave 2
982	Ahh Buzz Wave
983	OB Wave 1
984	OB Wave 2
985	OB Wave 3
986	Tenor tune alt
987	Dual Ride 1
988	Black Fills C
989	Orc Perc Preview
990	<GM>Standard Kit
991	<GM> Orch Kit
992	Castanets x 3
993	Tambourine x 3
994	Black Fills B
995	Black Fills A
996	2HandDrumCrp NB
997	Sleigh Loop
998	BD Rumble <V2.0>
999	Church Bell

Samples

900	Oboe
901	English Horn
902	Bassoon
903	Clarinet
904	Dbl Reeds
910	SoftTrump
911	French Horn
912	FrenchHrnSect
913	Tuba
914	Synth Accord
915	Tuba % Horn
920	Timp
921	Snare Roll
922	Snare Hit
923	Orch Bass
924	Orch Crash
925	Tam Tam
926	Triangle
927	Tamb Roll
928	Tamb Hit
929	Sleigh Bells
930	Woodblock
931	Low Clave
932	Castanet Hit
933	Castanet Up
934	Bi TamTam<v2.0>
935	Orch Crash ignf
937	Dark Triangle
938	MuteTriangle
939	Triangle (rel)
944	Xylophone
945	Glockenspiel
946	Chimes
950	Celeste
951	Harp
953	Nylon String Gt
957	Acoustic Bass
960	Pizz Strings
962	Solo Violin
963	Solo Viola
964	Solo Cello
965	Fast Solo Cello
966	Solo Double Bass
967	Conga Tone ignrl
968	Amb Kick 3 va
980	Organ Wave 8
981	Buzz Wave 2
982	Ahh Buzz Wave
983	OB Wave 1
984	OB Wave 2
985	OB Wave 3
988	Jackhammer
989	Scratch
990	Zap 1
991	Alarm Bell
992	DeepHouseClave
993	ChinaCrash
994	Dry Side Stick
995	Med Open Hi Hat
996	Syn Vibra Stick
997	Sleigh Loop
998	BD Rumble <v2.0>
999	Church Bell

Program Control Assignments

The preset programs in the K2600 Orchestral ROM option are organized by category. You can either use them as they are or as a good starting point for your own work. There are many ways to put expressivity and variety in a single program by assigning controllers to the various DSP functions in its layers. This list describes how each of the preset programs can be modulated or altered by various controllers. Only those control assignments that may not be immediately evident are listed. Control assignments like attack velocity and keynumber apply to most programs.

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
Pianos					
788	Piano Trio		Ride cymbal fade	Vibrato - Bass	
789	Pno & Syn String	String fade	Stringswell		
790	Fluid Grand		Wet/Dry mix		
791	Haunted Piano	Harp balance	Wet/Dry mix		
792	Xylopiano	Release ctl	Wet/Dry mix		
793	Grand,Harp&Lead	Lead tremolo	Lead fade	Lead tremolo	Sustain pedal does not affect the lead sound
Orchestras					
900	TotalCntrl Orch1	Layer bal	Adds brass & flute, boosts strings	Swell (trp out - ww solo)	
901	TotalCntrl Orch2	Layer bal, adds harp	Layer balance, adds horns/ cuts woodwinds	Swell	
902	BaroqueOrchestra	None	None	Swell	Sost ped disables brass
903	Oboe&Flute w/Str	Strings fadeout	Disables strings	None	
904	Horn&Flute w/Str	Strings fadeout	Disables strings	None	
905	Trp&Horns w/Str	Strings fadeout	Disables strings	None	
Winds					
906	Piccolo	None	Wet/Dry mix	None	
907	Orchestral Flute	Envelope control (slower)	Wet/Dry mix	None	
908	Solo Flute	Timbre (brighter)	Wet/Dry mix	None	
909	Orchestral Oboe	Swell	Wet/Dry mix, rate & depth	Vibrato	
910	Solo Oboe	Vibrato off	Wet/Dry mix	Swell	
911	2nd Oboe	Vibrato off	Wet/Dry mix	Swell	
912	Orch EnglishHorn	Swell	Wet/Dry mix, rate & depth	Vibrato	
913	Solo EnglishHorn	Vibrato off	Wet/Dry mix	Swell	
914	Orch Clarinet	Swell	Wet/Dry mix	Vibrato depth	
915	Solo Clarinet	Swell	Wet/Dry mix	Swell	
916	Orch Bassoon	Swell	Wet/Dry mix	Vibrato depth	
917	Solo Bassoon	Vibrato off	Wet/Dry mix	Swell	
918	Woodwinds 1	None	Wet/Dry mix	None	
919	Woodwinds 2	None	Wet/Dry mix, rate & depth	Swell, vibrato	

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
Brass					
920	Dynamic Trumpet	Swell	Wet/Dry mix	Vibrato depth	
921	Copland Sft Trp	Vibrato off	Wet/Dry mix	Swell	
922	Orch Trumpet	Timbre (darker)	Envelope Control	Swell, vibrato rate & depth	
923	Soft Trumpet	None	Wet/Dry mix	Vibrato depth	
924	Strght Mute Trp	Vibrato off	Wet/Dry mix	Swell	
925	French Horn MW	Timbre (brighter)	Wet/Dry mix	Vibrato rate & depth	
926	Slow Horn	Vibrato	Wet/Dry mix	None	
927	F Horn Con Sord	Timbre (brighter)	Wet/Dry mix	Vibrato depth	
928	F Horn a2 MW	Timbre (brighter)	Wet/Dry mix	None	
929	French Horn Sec1	None	Wet/Dry mix	Slight swell	
930	French Horn Sec2	None	Wet/Dry mix	Swell	
931	Solo Trombone	Selects legato layer	Wet/Dry mix	Slight swell when MW is off	
932	Tuba	Vibrato rate & depth	Wet/Dry mix	Vibrato rate & depth	
933	Dyn Hi Brass	Swell, legato	Wet/Dry mix	Swell	
934	Dyn Lo Brass	Swell, legato	Wet/Dry mix	Swell	
935	Dyn Brass & Horn	Timbre (darker)	Wet/Dry mix	None	
936	Soaring Brass	None	Wet/Dry mix	None	
937	MarcatoViolin MW	Spiccato articulation	Wet/Dry mix	Vibrato rate & depth	
938	Solo Violin	Delays auto-vibrato	Wet/Dry mix	Vibrato rate & depth	
939	2nd Violin	Envelope control	Wet/Dry mix	Vibrato rate	
940	Orch Viola	Release time (shorter)	Wet/Dry mix	Vibrato depth	
941	Solo Viola	Delays auto-vibrato	Wet/Dry mix	Vibrato rate & depth	
942	Slow Viola	Timbre (darker)	Wet/Dry mix	Swell, vibrato rate & depth	
943	MarcatoCello MW	Spiccato articulation	Wet/Dry mix	Vibrato rate & depth	
944	Solo Cello	Delays auto-vibrato	Wet/Dry mix	Vibrato rate & depth	
945	Slow Cello	Timbre (brighter)	Wet/Dry mix	Vibrato rate, swell	
946	Arco Dbl Bass	Bass boost	Wet/Dry mix	Vibrato depth	
947	Slow Arco Bass	Delays auto-vibrato	Wet/Dry mix	Swell, vibrato rate & depth	
948	Brt Dbl Bass	Decrescendo	Wet/Dry mix	Vibrato rate	
Section Strings					
949	Touch Strings	Timbre (brighter)	Envelope Control	Swell	
950	Fast Strings MW	Selects faster strings	Timbre (darker), Wet/Dry mix	Swell	
951	Chamber Section	None	Wet/Dry mix	Vibrato depth	
952	Sfz Strings MW	Tremolo	None	Swell	
953	Sweet Strings	Fade out	Wet/Dry mix	Vibrato depth	
954	Baroque Strg Ens	Bass boost, layer delay	Wet/Dry mix	Swell	
955	Big String Ens	None	Wet/Dry mix	Swell	
956	Bass String Sec	Bass boost on solo layer	Wet/Dry mix	None	
957	Pizzicato String	Timbre (darker)	Wet/Dry mix	None	
958	Wet Pizz	Treble boost	Wet/Dry mix	None	
959	Arco & Pizz	Timbre (brighter), layer balance	Enables 2nd string layer, stereo panning	Swell	

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
Plucked Strings					
960	Classical Guitar	Fade/disables key-up layer	Wet/Dry mix	None	
961	Virtuoso Guitar	Vibrato rate & depth	Wet/Dry mix	None	Sost ped enables stacato envelope
962	Acoustic Bass	Vibrato rate & depth	Wet/Dry mix	None	
963	Snappy Jazz Bass	Vibrato rate & depth	Pitch of snap, disables ride	Vibrato rate & depth	Sost ped disables ride cymbal
964	Dynamic Harp	Release time (longer)	Wet/Dry mix	None	
965	Harp w/8ve CTL	Brightness	Enables octave	None	
966	Harp Arps	None	Selects diminished	None	
Keyboards					
967	Celesta	None	Wet/Dry mix	None	
968	Pipes	Timbre (hollow)	Wet/Dry mix	None	
969	Pedal Pipes	None	None	None	
970	Church Bells	Distance	Timbre (brighter)	None	
Percussion					
971	Glockenspiel	None	Wet/Dry mix	None	Sus ped enables key-up layer (for rolls)
972	Xylophone	Timbre (fuller)	Wet/Dry mix	None	Sus ped enables key-up layer (for rolls)
973	Chimes	None	Wet/Dry mix	None	
974	Timpani/Chimes	Alt attack (timp)	Wet/Dry mix	None	
975	Timpani	Alt attack	Wet/Dry mix	None	Sus ped enables key-up layer (for rolls)
976	Timpani & Perc	Alt attack (timp)	None	None	Sost ped enables bass drum. Sus ped dampens.
977	Big Drum Corp	None	Enables both fill layers (black keys: f#3-a#4)	None	Sost ped switches layers. Sus ped dampens.
978	Orch Percussion1	None	Switches fill layers	None	Sus ped dampens
979	Orch Percussion2	None	Wet/Dry mix	None	Sus ped dampens
980	Jam Corp	Alt attack	Pitch control (black keys: f#3-a#4)	None	
981	Conga & Perc	Pitch control	Wet/Dry mix	None	
982	Woody Jam Rack	Pitch control up to 1200ct	Enables random drum layer	None	
983	Metal Garden	Pitch control up to 1200ct	Pitch control down to -1200ct	None	
984	Hot Tamali Kit	Tunes drums, alt atk on snares	Switches to old drum map	None	
985	Funk Kit	Tunes drums	Switches to old drum map	None	

Appendix F
Live Mode Objects

Live Mode Programs

740	LM VirtualDesk 1
741	LM VirtualDesk 2
742	LM EQ Room Hall
743	LM TubeAmp+ Gtr
744	LM Synth Sliders
745	LM EQ StIm Hall
746	LM ParaFlange
747	LM EQ Overload
748	LM Filters
749	LiveMode Default

The objects listed in this Appendix are current with operating system version 1.01. Your K2600 probably has version 1.01 objects installed. Here's how you can check the version of the objects you have installed:

1. Press the **Master** mode button to enter Master mode.
2. Select the Intonation parameter.
3. Change its value to **21**. You should see something like this: **21 Obj L1.00**. This "intonation table" is actually the version number of the K2600's basic object file (hence the **L**). If you scroll higher in the list, you may see other version numbers, depending on the ROM block options you have.

If your instrument doesn't have version 1.01 objects, you can get them from our website:

http://www.youngchang.com/kurzweil/html/downloads.html

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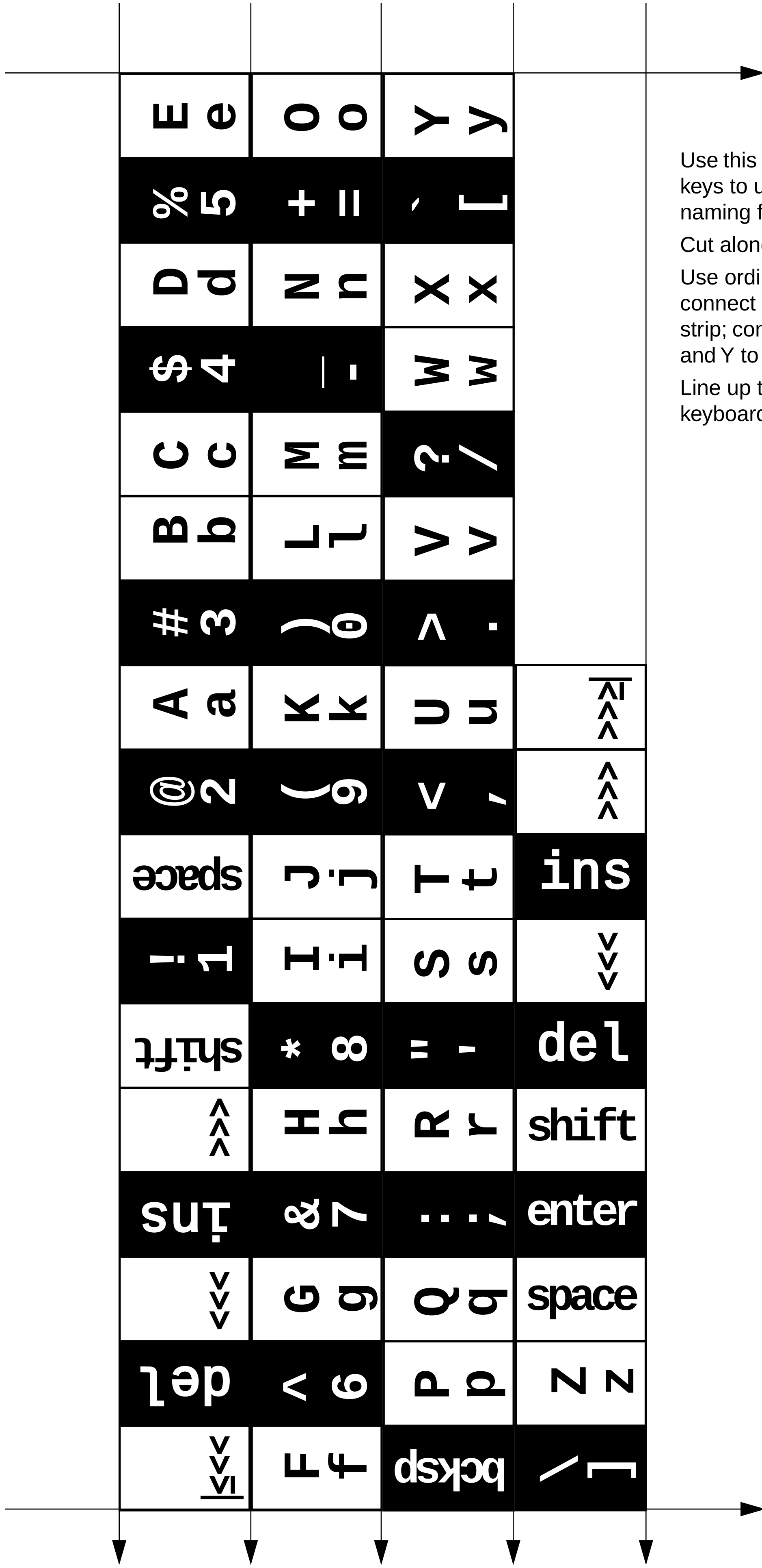

Use this chart to help you learn the keys to use for the keyboard naming feature.

Cut along the arrows as indicated.

Use ordinary transparent tape to connect the pieces into one long strip; connect E to F, O to backsp, and Y to].

Line up the strip with your keyboard so that A aligns with A 2.