K2661

Musician's Reference

Part Number: 910400 Rev. A

CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK,
DO NOT REMOVE THE COVER
NO USER SERVICEABLE PARTS INSIDE
REFER SERVICING TO QUALIFIED SERVICE PERSONNEL

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 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 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

Important Safety Instructions

1) Read these instructions
2) Keep these instructions.
3) Heed all warnings.
4) Follow all instructions.
5) Do not use this apparatus near water.
6) Clean only with dry cloth.
7) Do not block any of the ventilation openings. Install in accordance with the manufacturer's instructions.
8) Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
9) Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.

10) Protect the power cord from being walked on or pinched, particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
11) Only use attachments/accessories specified by the manufacturer.
12) Use only with a cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
13) Unplug this apparatus during lightning storms or when unused for long periods of time.
14) Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

Warning- To reduce the risk of fire or electric shock, do not expose this apparatus to rain or moisture. Do not expose this equipment to dripping or splashing and ensure that no objects filled with liquids, such as vases, are placed on the equipment.

To completely disconnect this equipment from the AC Mains, disconnect the power supply cord plug from the AC receptacle.

Kurzweil International Contacts

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

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Fax: (253) 984-0245

Young Chang Canada Corp.
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Official distributors in other countries are listed on the web site.

World Wide Web Home Page:

http://www.kurzweilmusicsystems.com

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Chapter 1
Front Panel

Front Panel Quick Reference

This section describes the features of the front panel of your K2661.

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.

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 (see the *Musician's Guide*.).

The Display

You may want to adjust the contrast and brightness of the display for different lighting conditions. There are two adjustment knobs on the rear panel of the K2661.

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 K2661'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.

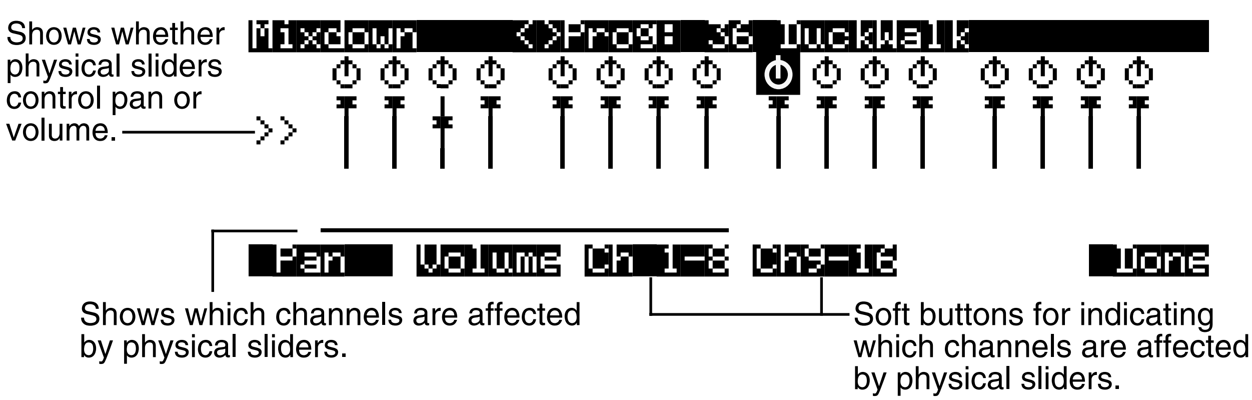
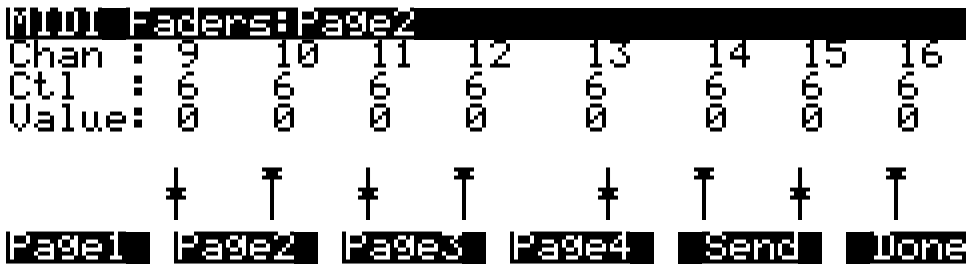


Figure 1-1 Mixdown Control

MIDI Faders button

When you press the **MIDI Faders** button, the K2661'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 K2661'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 1-1 describes all of the special button functions.

Button	**Mode or Editor**			
White Orange Light Grey	**Program Editor (Orange)**	**Setup Editor (Orange)**	**Song Mode**	**Sample Editor (Light Grey)**
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	
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 1-1 Special Button Functions

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.

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 1-2 Double Button Presses

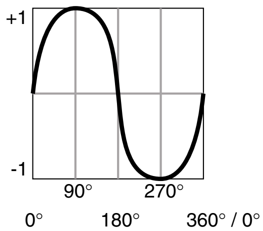
Chapter 2
LFOs

LFO Shapes

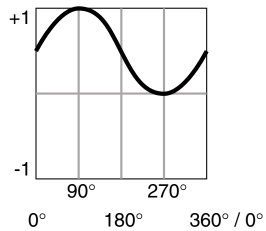
LFO Shape	Displayed As
Sine	Sine
Positive Sine	+Sine
Square	Square
Positive Square	+Squar
Triangle	Triang
Positive Triangle	+Trian
Rising Sawtooth	Rise S
Positive Rising Sawtooth	+Rise
Falling Sawtooth	Fall S
Positive Falling Sawtooth	+Fall
3 Step	3 Step
Positive 3 Step	+3 Ste
4 Step	4 Step
Positive 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

LFO Shapes

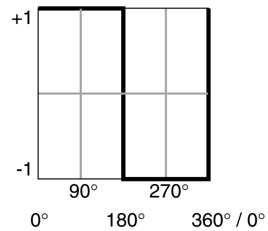
Sine



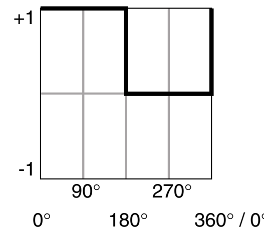
Positive Sine



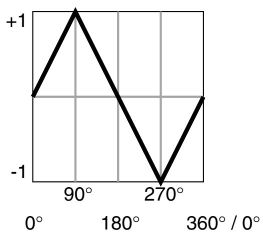
Sq uare



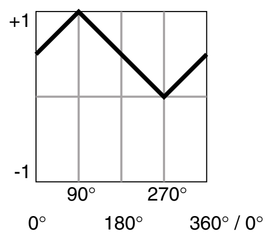
Positive Sq uare



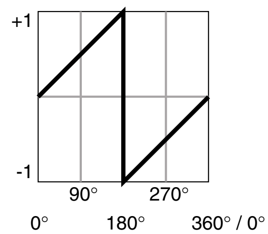
Triangle



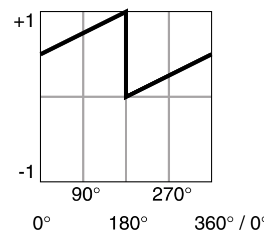
Positive Triangle



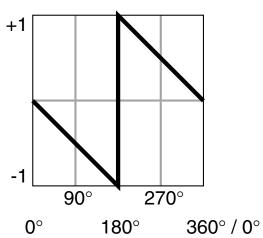
Rising Sawtooth



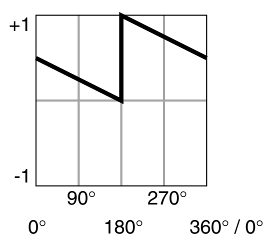
Positive Rising Sawtooth



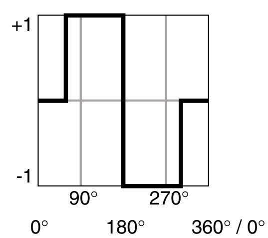
Falling Sawtooth



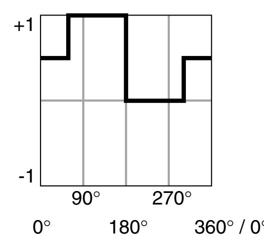
Positive Falling Sawtooth



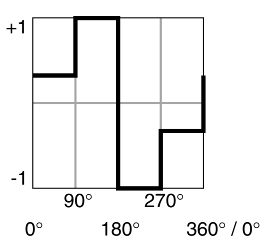
3 Step



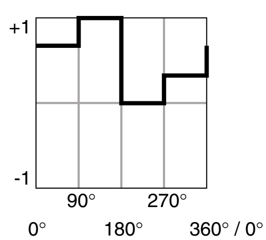
Positive 3 Step



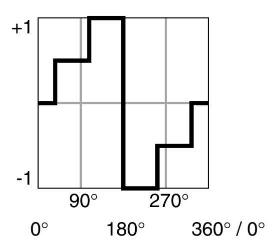
4 Step



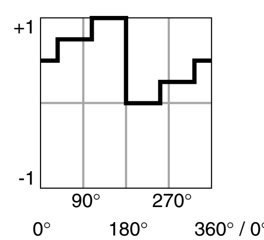
Positive 4 Step



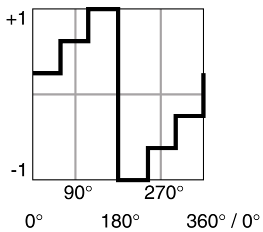
5 Step



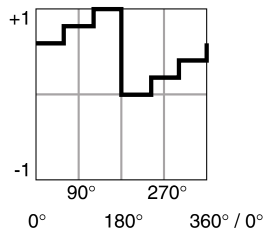
Positive 5 Step



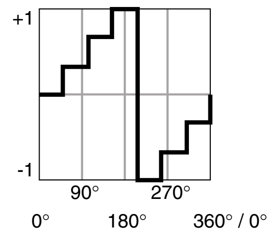
6 Step



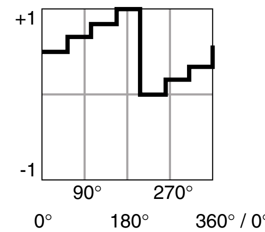
Positive 6 Step



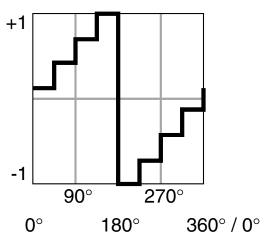
7 Step



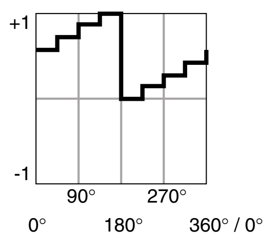
Positive 7 Step



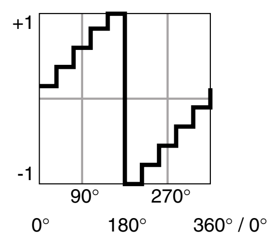
8 Step



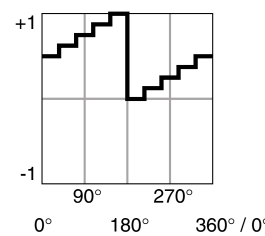
Positive 8 Step



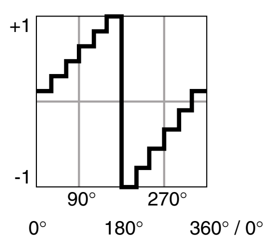
10 Step



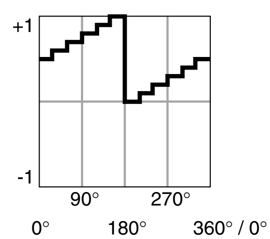
Positive 10 Step



12 Step



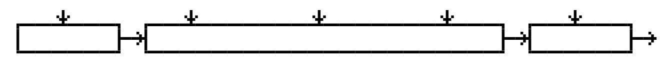
Positive 12 Step



Chapter 3
DSP Algorithms

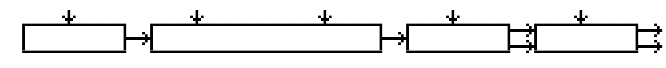
Note: Triple Mode algorithms are described in Chapter 12.

Algorithm 1



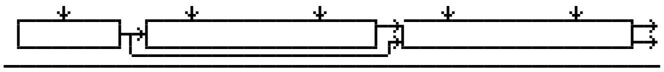
PITCH	HIFREQ STIMULATOR	AMP
	PARAMETRIC EQ	
	STEEP RESONANT BASS	
	4POLE LOPASS W/SEP	
	4POLE HIPASS W/SEP	
	TWIN PEAKS BANDPASS	
	DOUBLE NOTCH W/SEP	
	NONE	

Algorithm 2



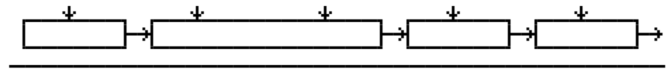
PITCH	2PARAM SHAPER	PANNER	AMP
	2POLE LOWPASS		
	BANDPASS FILT		
	NOTCH FILTER		
	2POLE ALLPASS		
	PARA BASS		
	PARA TREBLE		
	PARA MID		
	NONE		

Algorithm 3



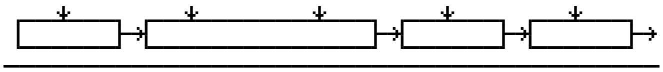
PITCH	2PARAM SHAPER	AMP U AMP L
	2POLE LOWPASS	BAL AMP
	BANDPASS FILT	
	NOTCH FILTER	
	2POLE ALLPASS	
	NONE	

Algorithm 4



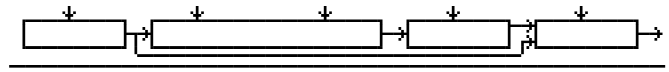
PITCH	2PARAM SHAPER	LPCLIP	AMP
	2POLE LOWPASS	SINE+	
	BANDPASS FILT	NOISE+	
	NOTCH FILTER	LOPASS	
	2POLE ALLPASS	HIPASS	
	PARA BASS	ALPASS	
	PARA TREBLE	GAIN	
	PARA MID	SHAPER	
	NONE	DIST	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 5



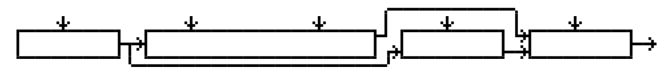
PITCH	2PARAM SHAPER	LP2RES	AMP
	2POLE LOWPASS	SHAPE2	
	BANDPASS FILT	BAND2	
	NOTCH FILTER	NOTCH2	
	2POLE ALLPASS	LOPAS2	
	PARA BASS	HIPAS2	
	PARA TREBLE	LPGATE	
	PARA MID	NONE	
	NONE		

Algorithm 6



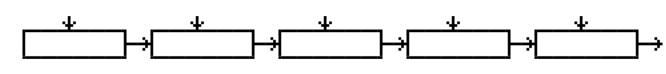
PITCH	2PARAM SHAPER	LPCLIP	x AMP
	2POLE LOWPASS	SINE+	+ AMP
	BANDPASS FILT	NOISE+	! AMP
	NOTCH FILTER	LOPASS	
	2POLE ALLPASS	HIPASS	
	NONE	ALPASS	
		GAIN	
		SHAPER	
		DIST	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 7



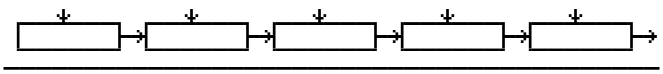
PITCH	2PARAM SHAPER	LPCLIP	x AMP
	2POLE LOWPASS	SINE+	+ AMP
	BANDPASS FILT	NOISE+	! AMP
	NOTCH FILTER	LOPASS	
	2POLE ALLPASS	HIPASS	
	NONE	ALPASS	
		GAIN	
		SHAPER	
		DIST	
		SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 8



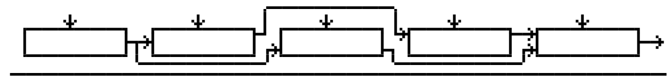
PITCH	LOPASS	LOPASS	LPCLIP	AMP
	HIPASS	HIPASS	SINE+	
	ALPASS	ALPASS	NOISE+	
	GAIN	GAIN	LOPASS	
	SHAPER	SHAPER	HIPASS	
	DIST	DIST	ALPASS	
	PWM	SW+SHP	GAIN	
	SINE	SAW+	SHAPER	
	LF SIN	WRAP	DIST	
	SW+SHP	NONE	SW+SHP	
	SAW+		SAW+	
	SAW		SW+DST	
	LF SAW		NONE	
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 9



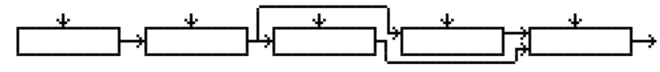
PITCH	LOPASS	LOPASS	LP2RES	AMP
	HIPASS	HIPASS	SHAPE2	
	ALPASS	ALPASS	BAND2	
	GAIN	GAIN	NOTCH2	
	SHAPER	SHAPER	LOPAS2	
	DIST	DIST	HIPAS2	
	PWM	SW+SHP	LPGATE	
	SINE	SAW+	NONE	
	LF SIN	WRAP		
	SW+SHP	NONE		
	SAW+			
	SAW			
	LF SAW			
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 10



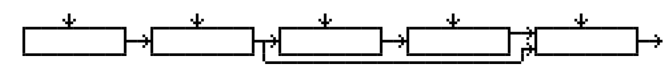
PITCH	LOPASS	LOPASS	LPCLIP	x AMP
	HIPASS	HIPASS	SINE+	+ AMP
	ALPASS	ALPASS	NOISE+	! AMP
	GAIN	GAIN	LOPASS	
	SHAPER	SHAPER	HIPASS	
	DIST	DIST	ALPASS	
	PWM	SINE	GAIN	
	SINE	LF SIN	SHAPER	
	LF SIN	SW+SHP	DIST	
	SW+SHP	SAW+	SW+SHP	
	SAW+	SAW	SAW+	
	SAW	LF SAW	SW+DST	
	LF SAW	SQUARE	NONE	
	SQUARE	LF SQR	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

Algorithm 11



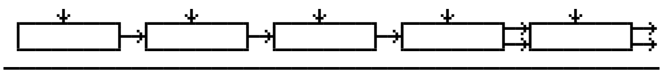
PITCH	LOPASS	LOPASS	LPCLIP	x AMP
	HIPASS	HIPASS	SINE+	+ AMP
	ALPASS	ALPASS	NOISE+	! AMP
	GAIN	GAIN	LOPASS	
	SHAPER	SHAPER	HIPASS	
	DIST	DIST	ALPASS	
	PWM	SINE	GAIN	
	SINE	LF SIN	SHAPER	
	LF SIN	SW+SHP	DIST	
	SW+SHP	SAW+	SINE	
	SAW+	SAW	LF SIN	
	SAW	LF SAW	SW+SHP	
	LF SAW	SQUARE	SAW+	
	SQUARE	LF SQR	SW+DST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

Algorithm 12



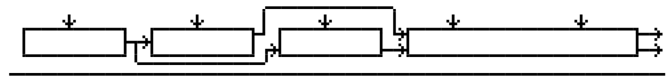
PITCH	LOPASS	LOPASS	LPCLIP	x AMP
	HIPASS	HIPASS	SINE+	+ AMP
	ALPASS	ALPASS	NOISE+	! AMP
	GAIN	GAIN	LOPASS	
	SHAPER	SHAPER	HIPASS	
	DIST	DIST	ALPASS	
	PWM	PWM	GAIN	
	SINE	SINE	SHAPER	
	LF SIN	LF SIN	DIST	
	SW+SHP	SW+SHP	SW+SHP	
	SAW+	SAW+	SAW+	
	SAW	SAW	SW+DST	
	LF SAW	LF SAW	NONE	
	SQUARE	SQUARE		
	LF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

Algorithm 13



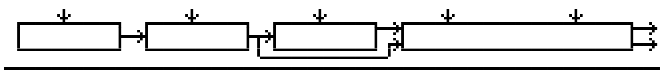
PITCH	LOPASS	LOPASS	PANNER	AMP
	HIPASS	HIPASS		
	ALPASS	ALPASS		
	GAIN	GAIN		
	SHAPER	SHAPER		
	DIST	DIST		
	PWM	SW+SHP		
	SINE	SAW+		
	LF SIN	WRAP		
	SW+SHP	NONE		
	SAW+			
	SAW			
	LF SAW			
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 14



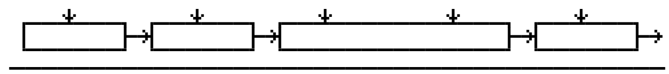
PITCH	LOPASS	LOPASS	AMP U AMP L
	HIPASS	HIPASS	BAL AMP
	ALPASS	ALPASS	
	GAIN	GAIN	
	SHAPER	SHAPER	
	DIST	DIST	
	SINE	SINE	
	LF SIN	LF SIN	
	SW+SHP	SW+SHP	
	SAW+	SAW+	
	SAW	SAW	
	LF SAW	LF SAW	
	SQUARE	SQUARE	
	LF SQR	LF SQR	
	WRAP	WRAP	
	NONE	NONE	

Algorithm 15



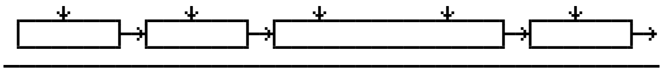
PITCH	LOPASS	LOPASS	AMP U AMP L
	HIPASS	HIPASS	BAL AMP
	ALPASS	ALPASS	
	GAIN	GAIN	
	SHAPER	SHAPER	
	DIST	DIST	
	PWM	SINE	
	SINE	LF SIN	
	LF SIN	SW+SHP	
	SW+SHP	SAW+	
	SAW+	SAW	
	SAW	LF SAW	
	LF SAW	SQUARE	
	SQUARE	LF SQR	
	LF SQR	WRAP	
	WRAP	NONE	
	NONE		

Algorithm 16



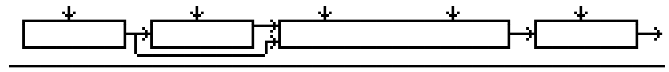
PITCH	LOPASS	PARA BASS	AMP
	HIPASS	PARA TREBLE	
	ALPASS	NONE	
	GAIN		
	SHAPER		
	DIST		
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		
	NONE		

Algorithm 17



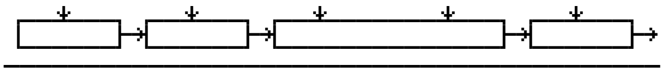
PITCH	LOPASS	SHAPE MOD OSC	AMP
	HIPASS	AMP MOD OSC	
	ALPASS	NONE	
	GAIN		
	SHAPER		
	DIST		
	PWM		
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		
	NONE		

Algorithm 18



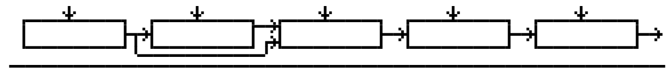
PITCH	LOPASS	x SHAPEMOD OSC	AMP
	HIPASS	+ SHAPEMOD OSC	
	ALPASS	NONE	
	GAIN		
	SHAPER		
	DIST		
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		
	NONE		

Algorithm 19



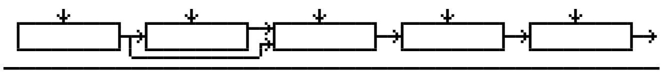
PITCH	LOPAS2	SHAPE MOD OSC	AMP
	NONE	NONE	

Algorithm 20



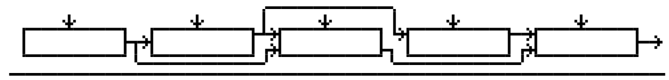
PITCH	LOPASS	x GAIN	LPCLIP	AMP
	HIPASS	+ GAIN	SINE+	
	ALPASS	XFADE	NOISE+	
	GAIN	AMPMOD	LOPASS	
	SHAPER	NONE	HIPASS	
	DIST		ALPASS	
	SINE		GAIN	
	LF SIN		SHAPER	
	SW+SHP		DIST	
	SAW+		SW+SHP	
	SAW		SAW+	
	LF SAW		SW+DST	
	SQUARE		NONE	
	LF SQR			
	WRAP			
	NONE			

Algorithm 21



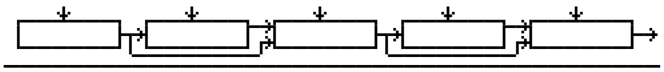
PITCH	LOPASS	x GAIN	LP2RES	AMP
	HIPASS	+ GAIN	SHAPE2	
	ALPASS	XFADE	BAND2	
	GAIN	AMPMOD	NOTCH2	
	SHAPER	NONE	LOPAS2	
	DIST		HIPAS2	
	SINE		LPGATE	
	LF SIN		NONE	
	SW+SHP			
	SAW+			
	SAW			
	LF SAW			
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 22



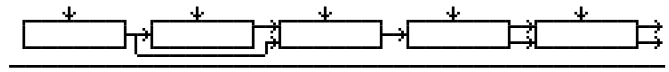
PITCH	LOPASS	x GAIN	LPCLIP	x AMP
	HIPASS	+ GAIN	SINE+	+ AMP
	ALPASS	XFADE	NOISE+	! AMP
	GAIN	AMPMOD	LOPASS	
	SHAPER	NONE	HIPASS	
	DIST		ALPASS	
	SINE		GAIN	
	LF SIN		SHAPER	
	SW+SHP		DIST	
	SAW+		SINE	
	SAW		LF SIN	
	LF SAW		SW+SHP	
	SQUARE		SAW+	
	LF SQR		SW+DST	
	WRAP		NONE	
	NONE			

Algorithm 23



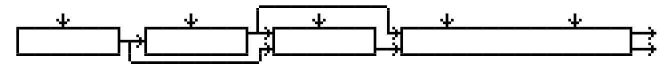
PITCH	LOPASS	x GAIN	LPCLIP	x AMP
	HIPASS	+ GAIN	SINE+	+ AMP
	ALPASS	XFADE	NOISE+	! AMP
	GAIN	AMPMOD	LOPASS	
	SHAPER	NONE	HIPASS	
	DIST		ALPASS	
	SINE		GAIN	
	LF SIN		SHAPER	
	SW+SHP		DIST	
	SAW+		SINE	
	SAW		LF SIN	
	LF SAW		SW+SHP	
	SQUARE		SAW+	
	LF SQR		SW+DST	
	WRAP		NONE	
	NONE			

Algorithm 24



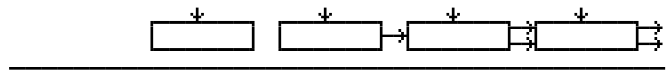
PITCH	LOPASS	x GAIN	PANNER	AMP
	HIPASS	+ GAIN		
	ALPASS	XFADE		
	GAIN	AMPMOD		
	SHAPER	NONE		
	DIST			
	SINE			
	LF SIN			
	SW+SHP			
	SAW+			
	SAW			
	LF SAW			
	SQUARE			
	LF SQR			
	WRAP			
	NONE			

Algorithm 25



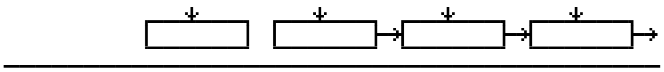
PITCH	LOPASS	x GAIN	AMP U AMP L
	HIPASS	+ GAIN	BAL AMP
	ALPASS	XFADE	
	GAIN	AMPMOD	
	SHAPER	NONE	
	DIST		
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		
	NONE		

Algorithm 26



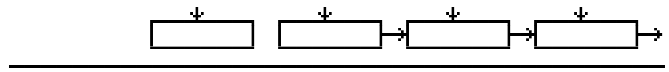
SYNC M	SYNC S	PANNER	AMP

Algorithm 27



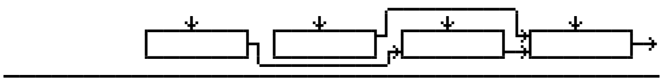
SYNC M	SYNC S	LPCLIP	AMP
		SINE+	
		NOISE+	
		LOPASS	
		HIPASS	
		ALPASS	
		GAIN	
		SHAPER	
		DIST	
		SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

Algorithm 28



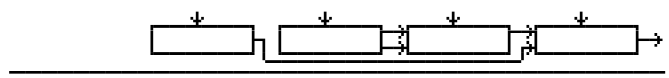
SYNC M	SYNC S	LP2RES	AMP
		SHAPE2	
		BAND2	
		NOTCH2	
		LOPAS2	
		HIPAS2	
		LPGATE	
		NONE	

Algorithm 29



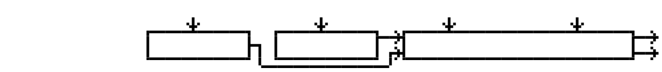
SYNC M	SYNC S	LPCLIP	x AMP
		SINE+	+ AMP
		NOISE+	! AMP
		LOPASS	
		HIPASS	
		ALPASS	
		GAIN	
		SHAPER	
		DIST	
		SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SW+DST	
		NONE	

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 K2661 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. The values you assign for these parameters determine which control messages will be transmitted to the K2661 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 K2661's sound engine, and, if the K2661's LocalKbdCh parameter matches my controller's transmit channel, also sends it to the K2661'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 K2661's sound engine, and will affect any control source parameter that has **Foot** assigned as its value.

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 K2661'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 K2661'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 K2661 by default.

5 **MIDI 05 (PortTim)**

This is the standard MIDI controller number for portamento time control. The K2661 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 K2661 interprets MIDI Controller 15 as AuxBend2. A value of 64 is centered.

16—19 MIDI 16—19 (Ctl A—D)

20 MIDI 20

21 MIDI 21 (AuxBend1)

The K2661 interprets MIDI Controller 21 as AuxBend1, which is assigned by default to the long ribbon (above the keyboard). 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 K2661 by default. The K2661 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 K2661 by default. The K2661 will always respond to sostenuto messages.

67 MIDI 67 (SoftPd)

This is the standard MIDI Controller number for Soft Pedal. The K2661 will always respond to Soft pedal messages.

68 MIDI 68

69 MIDI 69 (FrezPd)

The K2661 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 K2661 always responds to this message. When a MIDI Controller 75 message with a value above 64 is received, the K2661 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 K2661 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 K2661 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 K2661 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 in Chapter 6 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 in Chapter 6 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 in 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 K2661'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 K2661'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 K2661'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 Chapter 6 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 Chapter 6 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 Chapter 6 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 Chapter 6 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 Chapter 6 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 Chapter 14 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 K2661 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.

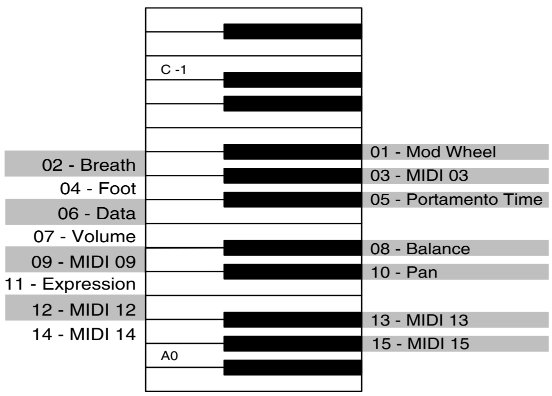
Assigned Value	Corresponding Constant	Assigned Value	Corresponding Constant
133	-0.99	201	0.09
134	-0.98	202	0.10
135	-0.97	203	0.12
136-140	-0.96 to -0.92	204	0.14
141	-0.91	205	0.16
142	-0.90	206-210	0.18 to 0.26
143-145	-0.88 to -0.84	211-215	0.28 to 0.36
146-150	-0.82 to -0.74	216-220	0.38 to 0.46
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.

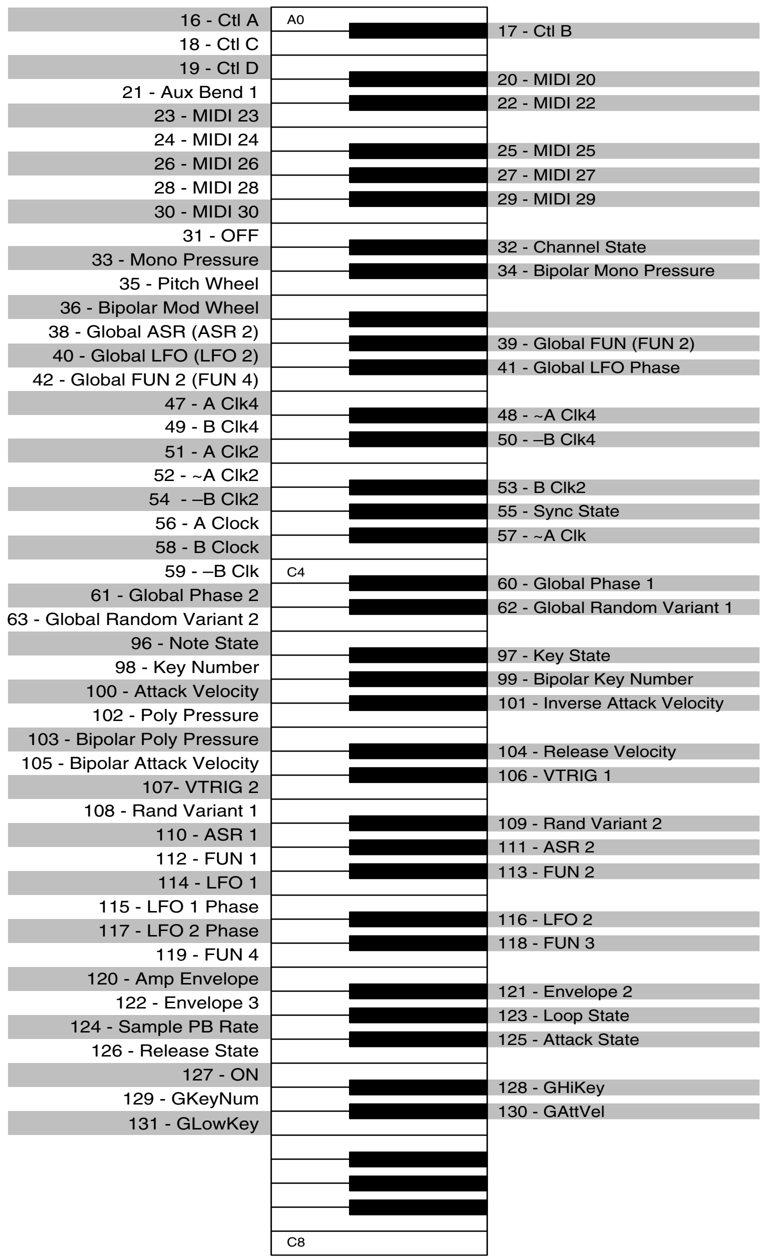
Keyboard Shortcuts for Control Sources

You can use the keyboard of your K2661 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

K2661 Note Numbers and MIDI Note Numbers

K2661	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 K2661 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 K2661 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 K2661's percussion programs have keymaps that place the various percussion timbres at standardized key locations. The K2661 includes the following drum keymaps: Preview Drums, five 5-octave kits (two dry and three ambient), a 2-octave kit, and the General MIDI kits. 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 K2661, as well as specific sections dealing with the Mac and the K2661.

Disk Size Restrictions

The K2661 can address up to 8 Gigabytes (8 G) of hard-disk space, in 2-G partitions. This is true for any hard disk formatted with the DOS-compatible FAT-16 format. Hard disks larger than 8 G can be formatted to make 8 G (in four partitions) accessible to the K2661. If you attach a *formatted* disk larger than 8 gigabytes, the K2661 may 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. For additional information, refer to:

http://www.kurzweilmusicsystems.com/html/scsi_help.html

http://www.kurzweilmusicsystems.com/html/drive_compatibility_info.html

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. The K2661 is peramanently terminated, so it will always act as an end device on the SCSI chain.

   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 K2661 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 K2661 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 (http://www.apstech.com, 1-800-395-5871) 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 K2661 and other devices first.

9. The K2661 file format is a proprietary format; no other device will be able to read or write a Kurzweil file.

10. The K2661 can read from and write to the first partition on a DOS-formatted disk.

11. 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 K2661 or another grounded metal object. Any devices connected to the SCSI bus should be turned off when plugging or unplugging SCSI cables.

    If the K2661 is connected to a Macintosh or PC you should make sure that the computer cannot access a SCSI disk at the same time the K2661 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 K2661. 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 K2661 access the disk at the same time there will be no damage to the hardware, but the bits on the disk, K2661, 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.

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

K2661 and Macintosh Computers

There are several points to consider when using a Macintosh with the K2661:

1. The Mac is not well equipped for having another SCSI master on the bus (that is, the K2661). It assumes that it owns the bus and its drives—consequently it will not allow the K2661 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 K2661 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 K2661.

2. The Mac and the K2661 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 K2661-formatted disks. To prevent problems, you will need to unmount the drive from the Mac desktop before using a K2661-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 K2661 at that moment.

3. The only good reason for connecting the Mac and the K2661 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 K2661 have its own drive, and the Mac have its own drive.

   However, we have discovered that when using a K2661 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 K2661, put the drive to sleep from the K2661. You can then change to a K2661-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 K2661 will tell you it can't read the disk, but the Mac will be able to.

The MIDI Sample Dump Standard

Samples can be transferred between the K2661 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 K2661 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 K2661, 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 K2661'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 K2661 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 K2661, 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 K2661 from an external source such as a computer or sampler, first connect the MIDI Out port of the sampler (or computer) to the K2661's MIDI In port, and connect the K2661'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 K2661'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 K2661's SysEx ID parameter to **0** or **1**. Alternatively, if you set the SysEx ID to **127**, the K2661 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 K2661 will use for storing the sample. The K2661 sample number is mapped from the destination sample number as follows:

Sample Number	K2661 ID
0	uses lowest unassigned ID between 200 and 999.
1-199	adds 200 to the ID (for example, 5 becomes 205 in the K2661.)
200-999	ID is the same number.

If the sample number maps to a number already assigned to a RAM sample in the K2661, the RAM sample will be deleted before the K2661 loads the new sample. The K2661 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 K2661, which reserves IDs 1-199 for ROM samples, which cannot be loaded or dumped. To get around this, the K2661 adds 200 to any numbers between 1 and 199. Therefore, if you want to load a sample into the K2661 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 K2661 Sample* on page 6-6 to learn how to use samples once they've been dumped to the K2661.

Getting a Sample into a Sample Editor from the K2661

Connect the MIDI ports of the K2661 and the computer/sampler in a MIDI loop as described for the Sampler/Computer to K2661 procedure above.

Access the computer software's "Get Sample" page (it might be called something different). As with loading a sample into the K2661, the K2661 adds 200 to dump request sample numbers between 1 and 199. K2661 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 K2661 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 K2661 from another K2661

Connect the MIDI ports of the two K2661s in a MIDI loop as described for the Sampler/Computer to K2661 procedure above.

On the source K2661, 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 K2661. The source K2661 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 K2661 as it is on the source K2661.

Dumping from the K2661 to a Sampler

This procedure is the same as dumping a sample from one K2661 to another. This will work only if the sampler supports the MIDI Sample Dump Standard.

Dumping a Sample from the K2661 to a MIDI Data Recorder

This can be accomplished by connecting the MIDI Out port of the K2661 to the MIDI In port of the MIDI Data Recorder. Go to the Sample Editor and select the K2661 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 K2661's MIDI mode LED will flash while the data transfer is in progress.

Loading a Sample into the K2661 from a MIDI Data Recorder

Connect the MIDI Out port of the Data Recorder to the MIDI In port of the K2661. Load the appropriate file containing the MIDI Sample Dump data into the Data Recorder, and send the file. The K2661's MIDI mode LED will flash during this procedure.

Accessing a New K2661 Sample

First, select the K2661 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 K2661, it will have the same ID as it did on the other K2661. 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 K2661 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 K2661. 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 K2661

There are two reasons a K2661 will not accept a MIDI Sample Dump. First, a dump will not be accepted if the destination sample number maps to a K2661 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 K2661. There may be samples in the K2661 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 K2661 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 K2661 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 K2661 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 K2661

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 K2661 dump sample ID 203, which would ordinarily dump a different sample from the one you intended, possibly causing the dump to fail. The K2661 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 K2661 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 K2661 to dump sample 205, ask for sample 204.

Some samples in the K2661 are copy-protected. These include all ROM samples and possibly some third-party samples. The K2661 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 K2661 from an external source (for example, a computer or a sampler). This button will also halt a sample dump from the K2661. The K2661 will ask for confirmation before it aborts the sample dump.

SMDI Sample Transfers

You can use the SMDI data transfer format (SMDI stands for SCSI Musical Data Interchange—pronounced *smiddy)* to transfer mono and stereo samples to and from the K2661. SMDI is parallel, not serial, so sample transfers can be made much faster than with the MIDI sample dump standard. Cycling '74's Max/MSP (http://www.cycling74.com) is a SMDI-capable Macintosh software package. Other applications that may be useful include:

Bias' Peak (http://www.bias-inc.com)

Mark Of The Unicorn's Digital Performer (http://www.motu.com)

Dissidents' Sample Wrench (http://www.dissidents.com)

Sonic Foundry's Sound Forge (http://www.sonicfoundry.com)

Propellerhead's ReCycle (http://www.propellerheads.se)

Applications such as these have commands for getting and sending samples, which is how you'll make the transfer from your offline storage to the K2661. Once the samples have been loaded to the K2661, you can use the Keymap and Sample Editors as you would with any other sample.

When transferring samples via SMDI, the K2661'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

K2661 System Exclusive Implementation

The MIDI System Exclusive capabilities of the K2661 allow you to manipulate objects in the K2661's memory from a computer system, another K2661, or a MIDI data recorder. The following is a reference to the SysEx protocol used by the K2661. 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 K2661 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 K2661, or a K2661 program to a K2000 via MIDI system exclusive.*

Common Format

In the following discussion, the fields of the K2661 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 K2661 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 K2661 will recognize a SysEx message if dev-id is the same is the SysEx ID parameter from the MIDI-mode RECEIVE page. If the K2661'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 K2661.

msg-type is the identifier of one of the K2661 SysEx messages defined below, and message is the variable-length message contents.

eox is always F7h, for end of System Exclusive.

Data Formats

K2661 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 K2661'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 K2661 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 K2661 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 K2661 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 K2661 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 K2661 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 K2661 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 K2661 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 K2661 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 K2661, which will add all of the objects contained in the dump to its own object database. Important: If the K2661 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 K2661 before sending the dump, so that the K2661 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 K2661 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 K2661 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 K2661 (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 K2661 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 K2661 as if the buttons were pressed at its front panel. The button codes are listed in tables beginning on page 7-7. The K2661 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		

Button	Code (hex)
Play	48
Stop	4A
Record	4B
Slider1	4E
Slider2	50
Slider3	52
Slider4	54
Slider5	56
Slider6	58
Slider7	5A
Slider8	5C
MIDI Faders	5E
Mixdown	60
Solo	62

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

ALLTEXT = 15h

…requests all text in the K2661'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 K2661 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 K2661 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 K2661. It should always rest on a hard flat surface—and on its rubber feet, not on the bottom panel.

Never block the ventilation openings; doing so can cause overheating that will seriously damage your K2661. To provide adequate ventilation, the rear panel should be at least four inches from any vertical surface. Try to minimize the amount of dust in the environment.

The K2661's RAM backup battery, along with any sample RAM, or ROM block options you may install, are the only user-serviceable parts in the K2661. The only part you should ever disassemble on your K2661 is the access panel on the bottom of the instrument—removing anything else will void your product warranty.

Cleaning Your K2661

It's a good idea to remove dust from your K2661 occasionally. You may also want to remove fingerprints. You can clean the K2661'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 K2661.

Battery Replacement

The K2661 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 K2661 warns of low battery voltage.

The battery in your K2661 will last for several years.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 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 K2661 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 K2661 and disconnect the power cable!*

3. Carefully place your K2661 upside down on a padded level surface, with the front of the instrument toward you. Use soft, sturdy foam under the ends of the instrument, to protect the wheels and sliders.
4. Locate the access panel. It's about 6 by 13 inches in size, slightly to the right of center, toward the back of the instrument.
5. Remove the screws that hold the access panel in place, 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.

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 K2661 responds by lighting each LED in sequence and then displaying something like the following.

```
K2661 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 K2661 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).

Maximizing Music and Minimizing Noise

Your K2661 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 K2661'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 K2661 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 K2661 into a stage or studio environment is to use the K2661's balanced audio outputs, and to be sure that the mixing board, amplifier, or other equipment receiving the K2661 audio signal has a balanced input circuit.

If you can't use the K2661 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 K2661'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 K2661, 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 K2661's balanced outputs. A 75-watt transformer is sufficient for the K2661.

Use the shortest possible cable, with the heaviest possible ground (shield) wire, to connect your K2661 to the mixing board or amplifier. This helps to reduce the potential difference between the chassis of the K2661 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 K2661'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 K2661's front panel. Smaller gear like drum machines and hardware sequencers can also cause interference.

Power Problems and Solutions

The K2661 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 K2661.

Very low line voltage or severe voltage dips are a problem for any computer-based instrument. When the K2661 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 K2661 (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 K2661 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 230 volts in Europe) because overheating or blown fuses may occur if you leave the K2661 at the lower setting and use it later at a normal voltage level.

Troubleshooting

If you're not seeing the proper display or hearing the sounds you expect, 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 and brightness if necessary (there are two knobs on the rear panel). 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 K2661 and to your sound system. You may want to switch your audio cables, unless you're sure they're functioning properly.
- Check that the K2661'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 K2661 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 any RAM objects you've created to disk or SmartMedia, and perform a hard reset. Do this by pressing the Master-mode button, followed by the **MAST2** soft button, then pressing the **Reset** soft button (at the lower right of the display). The K2661 will warn you about deleting everything (only RAM objects will be deleted). Press **Yes**. After a few seconds, the power-up display should appear.

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 K2661 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.

No Signal at ADAT In

To use ADAT In, the K2661's ADAT Out cable must also be connected to the sending device. Since the K2661 must be the "master," the other device(s) must "slave" to it.

Chapter 9
Upgrading Sample Memory

Program RAM vs. Sample RAM

If you're creating a lot of your own programs, and using samples loaded from disk, here are some things you should know. First of all, there's an important distinction between *sample* RAM and *program* RAM. Sample RAM refers to the SIMM (Single In-line Memory Module) installed in your K2661 specifically for storing sample data. This RAM is reserved exclusively for sample storage; nothing else is stored there. Sample RAM is volatile; that is, when you power down your K2661, the data stored there is immediately erased. That's why you have to load RAM samples every time you power up.

The amount of sample RAM in your K2661 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 K2661 (or that the K2661 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 amount of free program RAM is indicated at the right side of the top line of the display in Song mode and Disk mode.

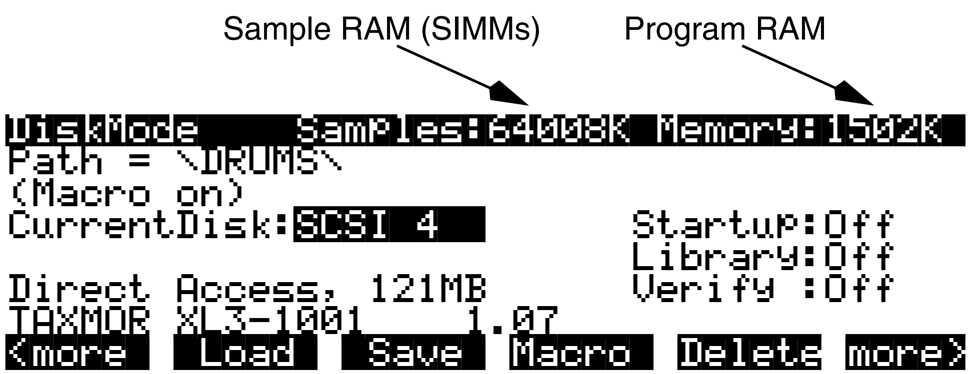


Figure 9-1 Disk mode page showing Sample RAM and Program RAM

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 or SmartMedia. This offers insurance in the unlikely event that the RAM becomes corrupted.

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 in Chapter 13 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 a SIMM for K2661 Sample Memory

If your K2661 has a 64 M SIMM installed, you can replace it with a 128 M SIMM to increase your 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 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)

There is space for a single SIMM, up to a total of 128 M.

These companies make SIMMs that work with the K2661 (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 K2661; 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 K2661, which you'll need to open to install your sample RAM. This is the same panel you would open to install a replacement battery or ROM sound block options.

***Warning**: Turn off your K2661 and disconnect the power cable!*

1. Carefully place your K2661 upside down on a padded level surface, with the front of the instrument toward you. Use soft, sturdy foam under the ends of the instrument, to protect the wheels and sliders.
2. Locate the access panel. It's about 6 by 13 inches in size, slightly to the right of center, toward the back of the instrument.
3. Remove the eight screws that hold the access panel in place, 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 socket for sample RAM.
5. Remove the installed SIMM and store it in a static-proof bag.
6. Place the new SIMM into the sample RAM socket. There's only one way that a SIMM will fit into the socket. Be sure the clips at the sides of the sockets snap into place.
7. Check the setting of the voltage jumper, and change it if it doesn't match the voltage of your SIMMs. The K2661 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 K2661 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.
8. 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.

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
138	Degen Regen BPM
139	Switch Loops
140	Moving Delay

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
161	Allpass Phaser 3
781	St Chorus+Delay
784	St Flange+Delay

Combination Algs

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
790	Gate+Cmp[EQ]+Vrb
792	Gate+TubeAmp

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 Algs

ID	Name
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
741	Rotor 1
742	VC+Dist+HiLoRot2
743	Subtle Distort
744	Quantize+Alias
745	Pitcher+Miniverb
746	Reverb+Compress

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
914	Reverse LaserVerb
915	Gated LaserVerb
916	Poly Pitcher
917	Frequency Offset
918	MutualFreqOffset
919	WackedPitchLFO
920	Chaos!
948	Band Compress
949	CompressDualTime
971	3 Band EQ
972	HF Stimulate 1
975	Harmonic Suppress

Studio / Mixdown Algs

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

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
104	Cool Dark Place
105	Gunshot Verb
106	Rvrb Compression
107	Snappy Drum Room
108	Roomitizer
109	Live To Tape
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
123	Rvrs Laserverb
124	Growler
125	Ringy Drum Plate
126	Oil Tank
127	Wobbly Plate
128	Pitcher Hall
129	Distant Pitch Room
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
139	Fanfare in Gmaj
140	Basic Delay 1/8
141	Diffuse Slaps
142	Multitaps ms
143	Timbre Taps
144	Ecko Plecks
145	Degenerator
146	Nanobot Feedback
147	Takes a while...
148	Wait for UFO
149	News Update
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
162	Full Chorus
163	Dense Gtr Chorus
164	Standard Gtr Chorus
165	Bass Chorus
166	StChorus+Delay
167	StChor+3vs2Delay
168	CDR for Lead Gtr
169	PinchChorusDelay
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
179	Soft Edge Flange
180	Ned Flangers
181	Wispy Flange
182	Crystal Flange
183	NarrowResFlange
184	TightSlapFlange
185	Flanged Taps
186	StFlange+Delay
187	StFlng+3vs2Delay
188	Singing Flanger
189	DampedEchoFlange
190	Circles
191	Slow Deep Phaser
192	Manual Phaser
193	Vibrato Phaser
194	ThunderPhaser
195	Saucepan Phaser
196	Static Phaser
197	Slippery Slope
198	Westward Waves
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
791	Slow Res Rotor
792	Smooth Rotors
793	Very Nazty Rotor
794	80's Funk Guitar
795	Mean 70'sFunkGtr
796	Crunch Guitar
797	Classic Gtr Dist
798	SaturatedGtrDist
799	TubeDist DlyChor
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
881	Chorus PanDelay
882	Flange + Delay
883	TubeAmp DlyChor
884	StChor+3vs2Delay
885	TubeAmp DlyChor2
886	Drum Crusher
887	Bass Env Filt 2
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
928	Gated Laserverb
929	Waterford
930	A little dirty
931	Slight Overload
932	Blown Speaker
933	Ring Linger
934	Drum Shaper
935	Aliaser
936	Quantize+Alias1
937	Quantize+Alias2
938	Drum Mortar
939	Superphasulate
940	Rich Noodle
941	Nickel Chorus
942	HF Stimulator
943	OddHarmSuppress
944	AM Radio
945	U-Shaped EQ
946	Drum Crusher
947	Vocal Room
948	Vocal Stage
949	Mid Compressor
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
981	Poly Pitcher
982	CheapVoxChanger
983	Hip Hop Aura
984	Woodenize
985	Marimbafication
986	Frequency Offset
987	Drum Loosener
988	Drum Tightener
989	Vox Honker
990	Glacial Canyon
991	Spring Thing
992	Contact
993	Drum Frightener
994	Mad Hatter
995	Fallout
996	Ascension
997	60Hz Buzz Kill
998	Stereo Analyze
999	FX Mod Diag

KDFX Studios

ID	Name
1	RoomChorDly Hall
2	RmChorChRv Hall
3	RoomChorCDR Hall
4	RoomChor Hall
5	RoomChrCh4T Hall
6	RoomFlngCDR Hall
7	RoomFlgEcho Hall
8	RmFlngStImg Garg
9	RmFlgChDly 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 Space
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 Hall2
47	Room Room Hall2
48	Room Hall Hall2
49	Sndbrd Room Hall
50	Sndbrd Rm Hall2
51	Room Room Hall3
52	auxChrMDly Room
53	auxFlngChRv Room
54	auxShp4MDly Hall
55	auxDistLasr Room
56	auxEnhSp4T Class
57	auxDistLasr Acid
58	EnhcManPhs Room
59	EnhrFlg8Tap 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 ChDly Hall
84	HallFlgChDl Hall
85	Hall Room SRS
86	Hall Room Room
87	Hall CmpRvb
88	Hall Flng Hall
89	HallRoomChr Hall
90	auxPhsrFDR Hall
91	auxChrDist+ Hall
92	auxFlgDist+ Hall
93	auxChrDst+ Hall
94	auxChorMDly Hall
95	auxChorSp6T Hall
96	auxChorChDl Hall
97	auxPhasStIm Hall
98	auxFlngCDR Hall
99	auxPhsrFldblHall
100	auxSRSRoom Hall
101	auxFlLsr SwHall
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 FlDly
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 CmpRv
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
198	Pre-KDFX Studio
199	Default Studio
700	Flanger Trio
701	Thin Cloud Layer
702	Fazetortion
703	Ultraphasulate
704	Chutney Squishy
705	BuddMeets Tomita
706	2 Buses 2 Chains
707	RingMod Envelope
708	Drum Megashaper
709	Dist Lead Guitar
710	Blues/R&R Guitar
711	Clean Rhythm Gtr
712	Sweeping Verb
713	Pad Ambience
714	Coming of Dawn
715	Wide Raindrops
716	Robot Voice
717	Before the Crash
718	Piano Multi-verb
719	Mastering Studio
749	E Pno Amp
750	Tron Effects
751	70s EfxDistB
752	Flange Tap Synth
753	HlFlgChDl Rm LD1
754	Hall CabDst Hall
755	2 Bus 2 Chain 2
756	Ster ChorusDelay
757	Phsr Trpflt
758	Jazzy Room
759	Drum Crusher
760	Verb Roto Room
761	Flt Cmp Ch4T
762	DrCmpr CDR
763	Chor Cmp Verb
764	Cmpr ChrDbl
765	ChorRotoCmp Rev
766	RevCmp Hall
767	Filt Chaos Room
768	SKComp MTChor
769	Qntflg Cmp Chor
770	SKCmp FlngVerb
771	CDR Cmp FlgHll
772	CDR Pan FlgHll
773	Verb RvCmp Delay
774	Chor&Flng Delays

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 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.

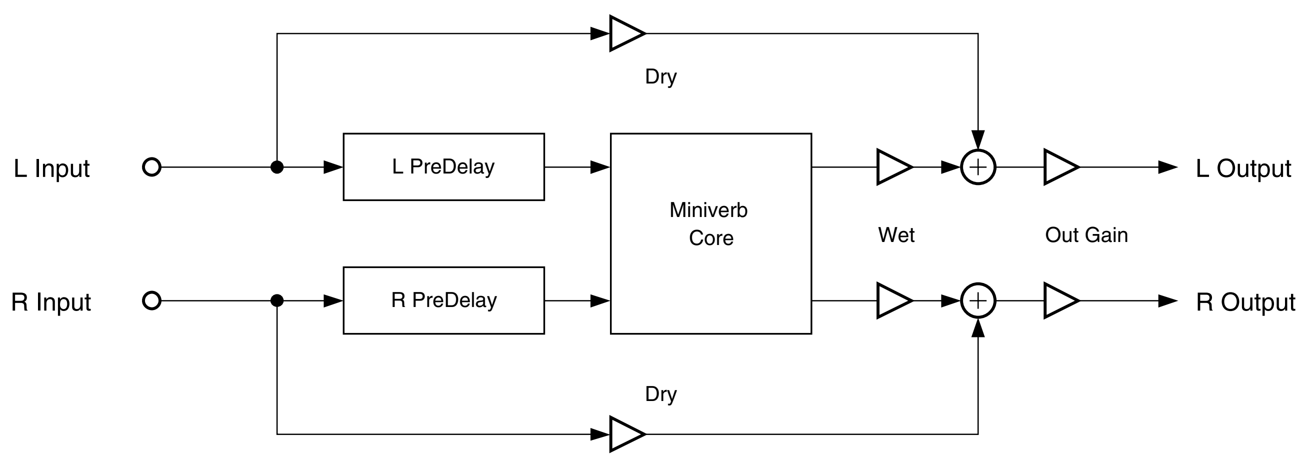


Figure 10-1 Simplified Block Diagram of MiniVerb

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.

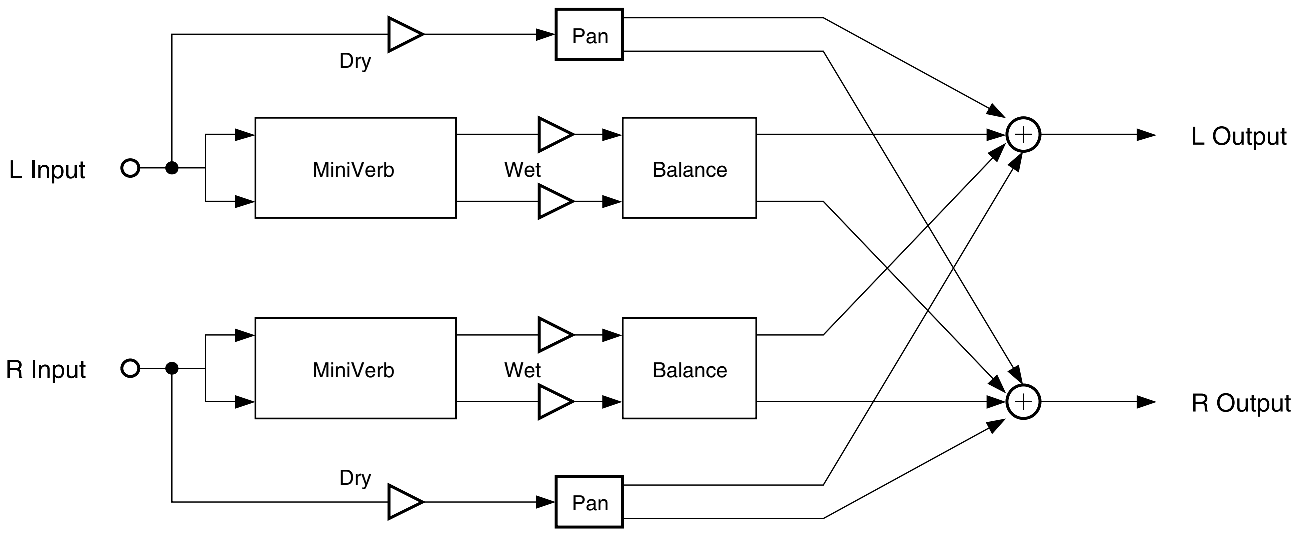


Figure 10-2 Simplified Block Diagram of Dual MiniVerb

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 behavior 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.

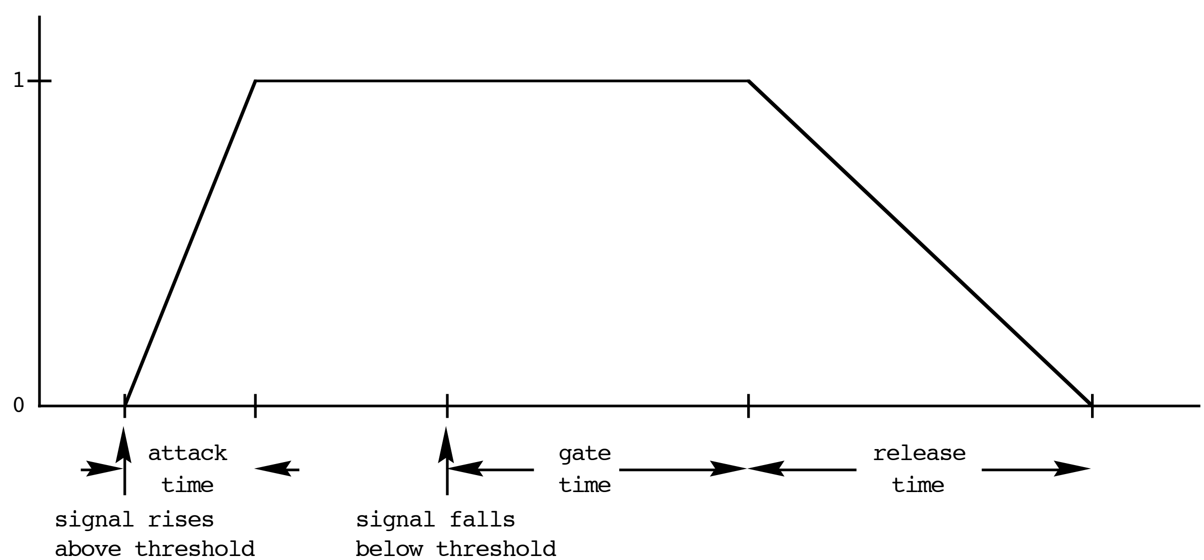


Figure 10-3 Gate Behavior

If Gate Duck is turned on, then the behavior 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.

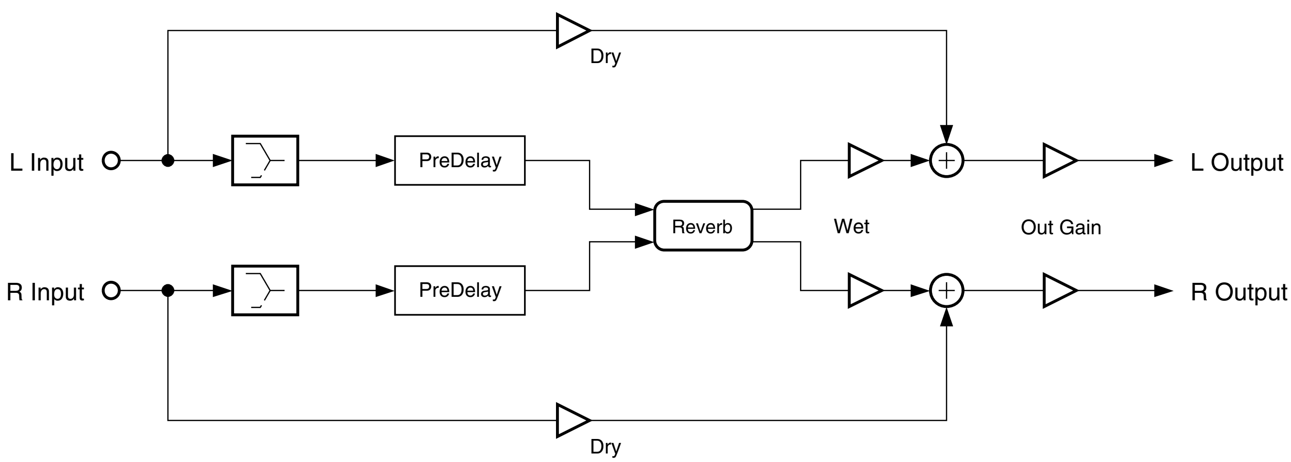


Figure 10-4 Simplified block diagram of Panaural Room.

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.

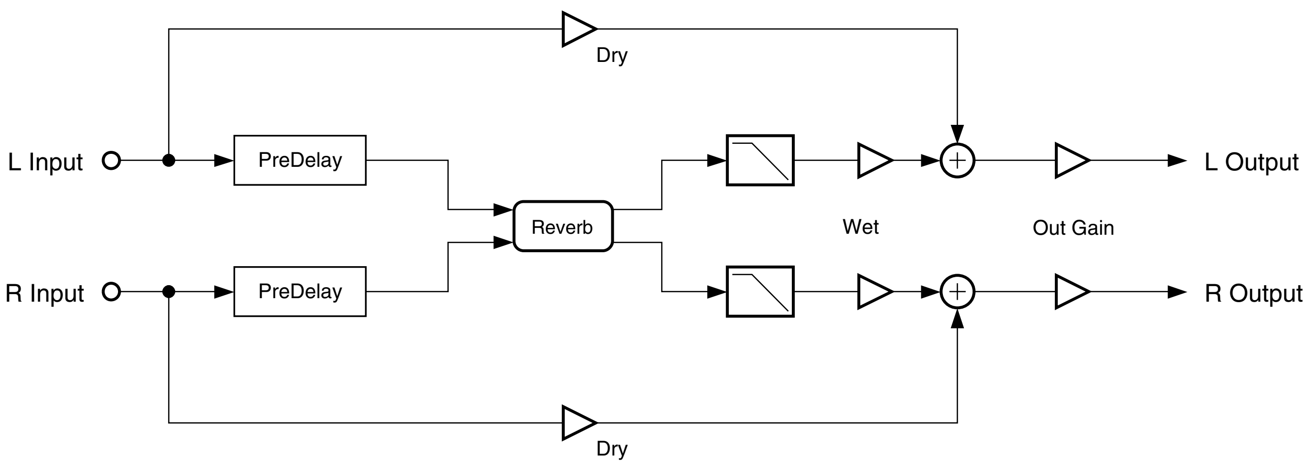


Figure 10-5 Simplified block diagram of Stereo Hall.

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 K2661.

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.

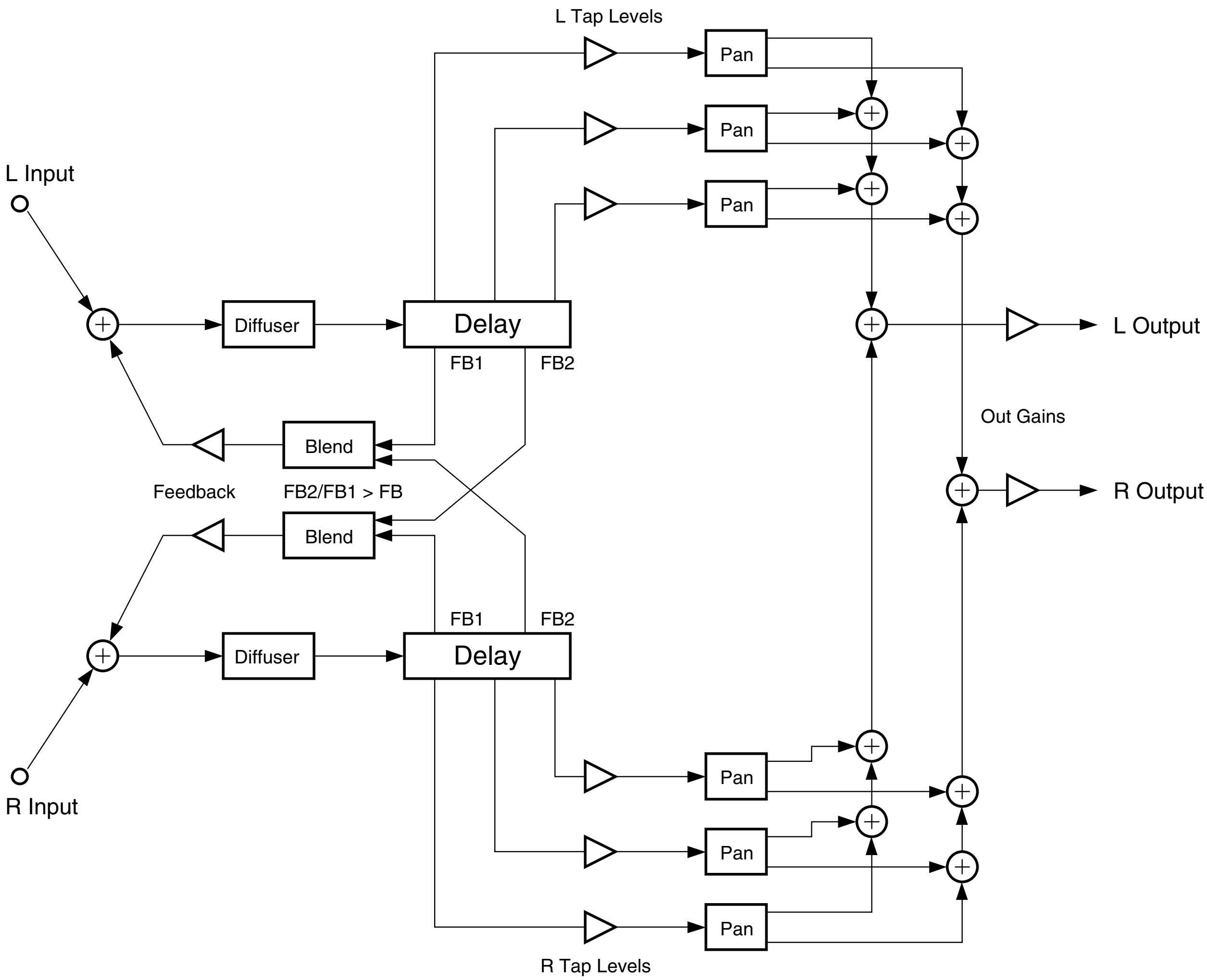

Figure 10-6 Signal flow of Complex Echo

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 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. 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.

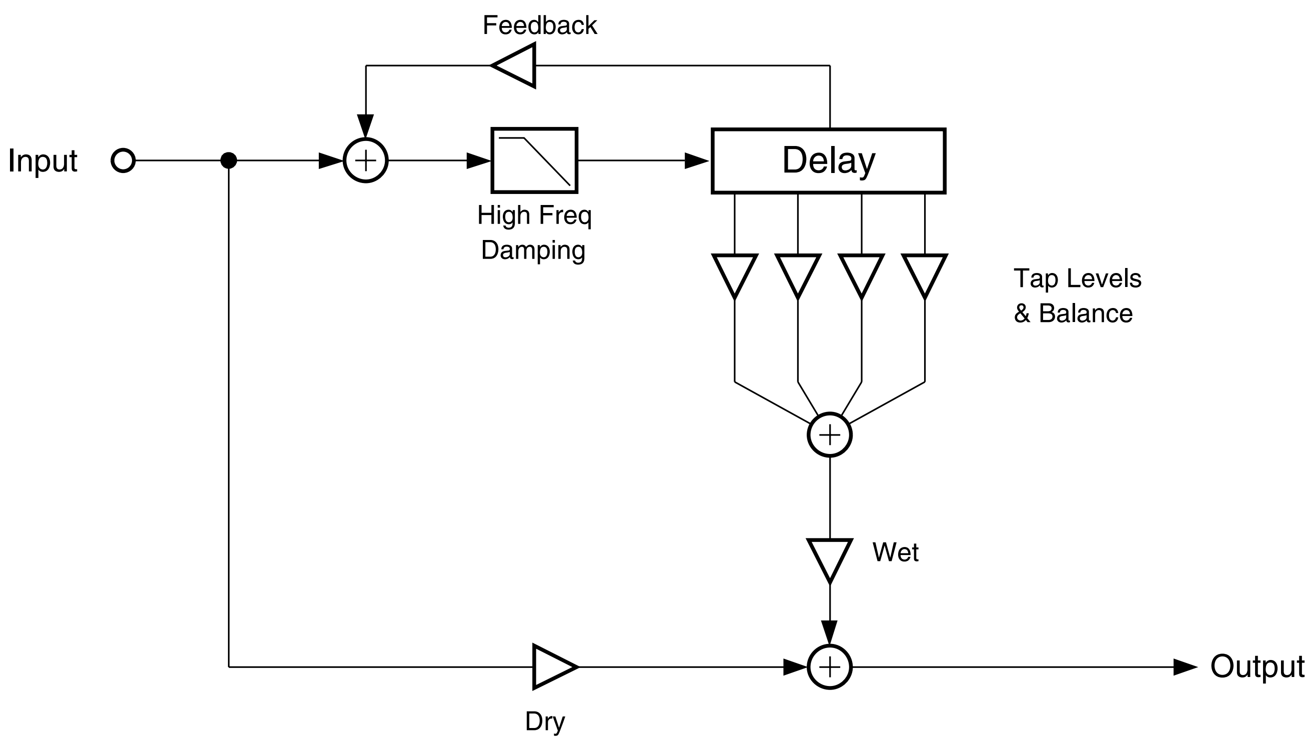

Figure 10-7 Left Channel of 4-Tap Delay

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.

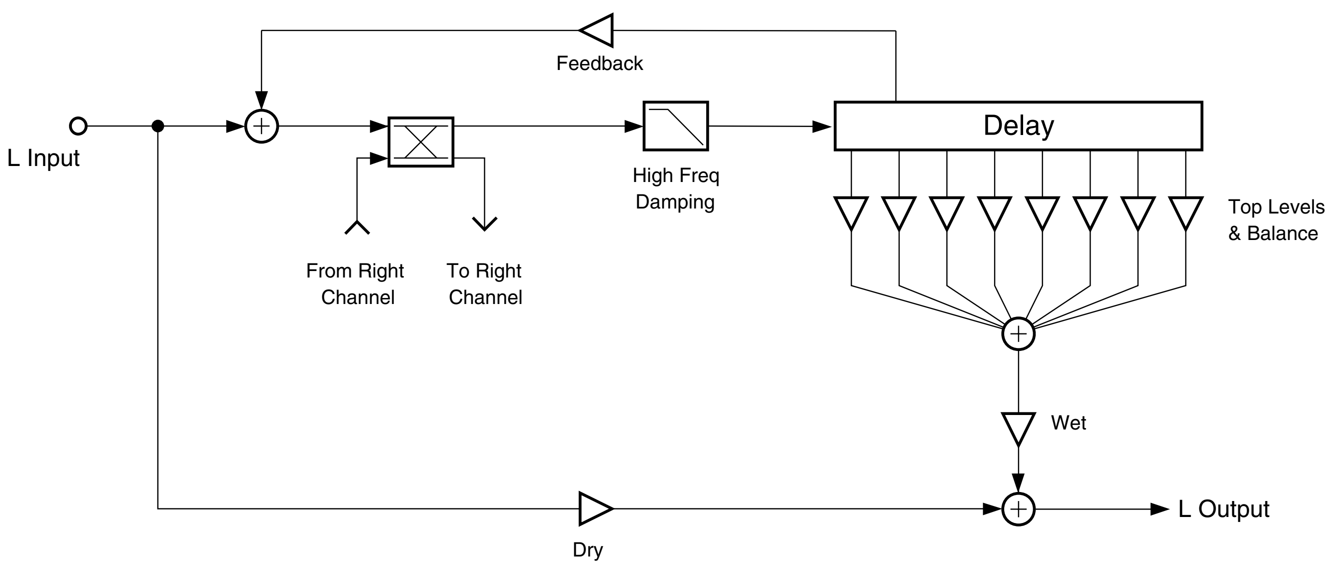

Figure 10-8 Left Channel of 8-Tap Delay

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

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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

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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

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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

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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.

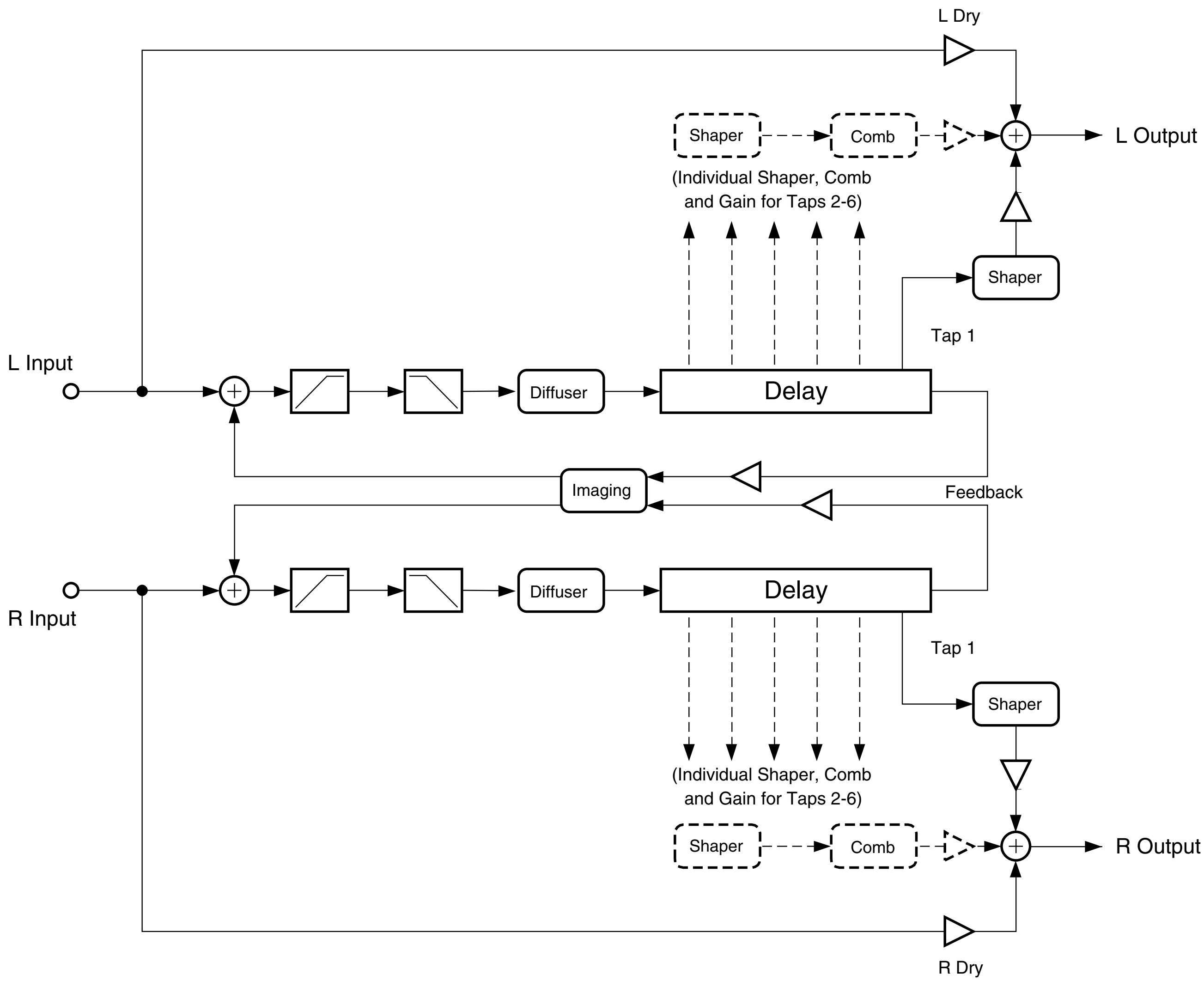

Figure 10-9 Spectral 6 Tap

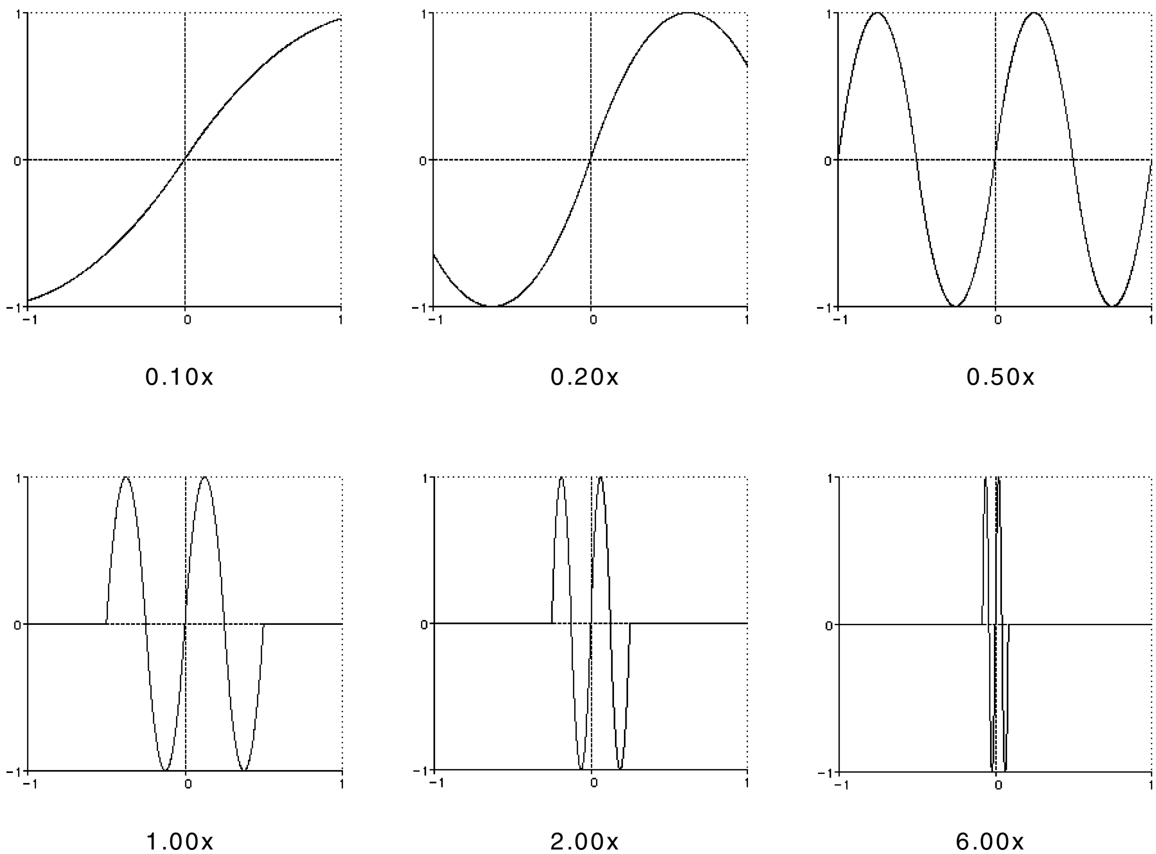

Figure 10-10 Various shaper curves used in the Spectral Multi-Taps

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 %

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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%

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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.

138 Degen Regen BPM

Long delay allowing loop instability

PAUs: 4 each

Degen Regen BPM starts as a simple mono delay line with feedback. However with the Fdbk Gain and Dist Drive parameters, the algorithm can be pushed hard into instability. When **Degen Regen BPM** is unstable, your sound gets a little louder on each pass through the delay line. Eventually the sound will hit digital clipping when the effects processor runs out of headroom bits. To keep this all under control, a soft-knee compressor has been included inside the delay line loop. With the compressor properly set, the sound never reaches digital clipping, but it does become more and more distorted as it gets pushed harder and harder into the compressor. To make things really nasty, there's also a distortion in the delay path. (The distortion parameters are on the PARAM4 page with the compressor parameters.)

Degen Regen BPM uses all 4 PAUs available for insert effects. With the resources of all 4 PAUs available, **Degen Regen BPM** lets you set the longest mono delay line available in KDFX which is just over 20 seconds. If you want a long delay, this is the algorithm to do it. (You don't have to over-drive the feedback or use the distortion.)

The delay has two output taps in addition to the feedback tap. Each tap may be moved along the delay line using an LFO (internal to the effects processor). The output taps have separate controls for level and panning (in the stereo configurations).

Throw a few filters into the delay line loop, and you get a pretty versatile delay line. The available filters are highpass (LF Damping), lowpass (HF Damping), bass shelf, treble shelf, and two parametric EQs (Mid1, Mid2).

For details about the compressor see the documentation for **SoftKneeCompress**. For the distortion see the documentation for **Mono Distortion**.

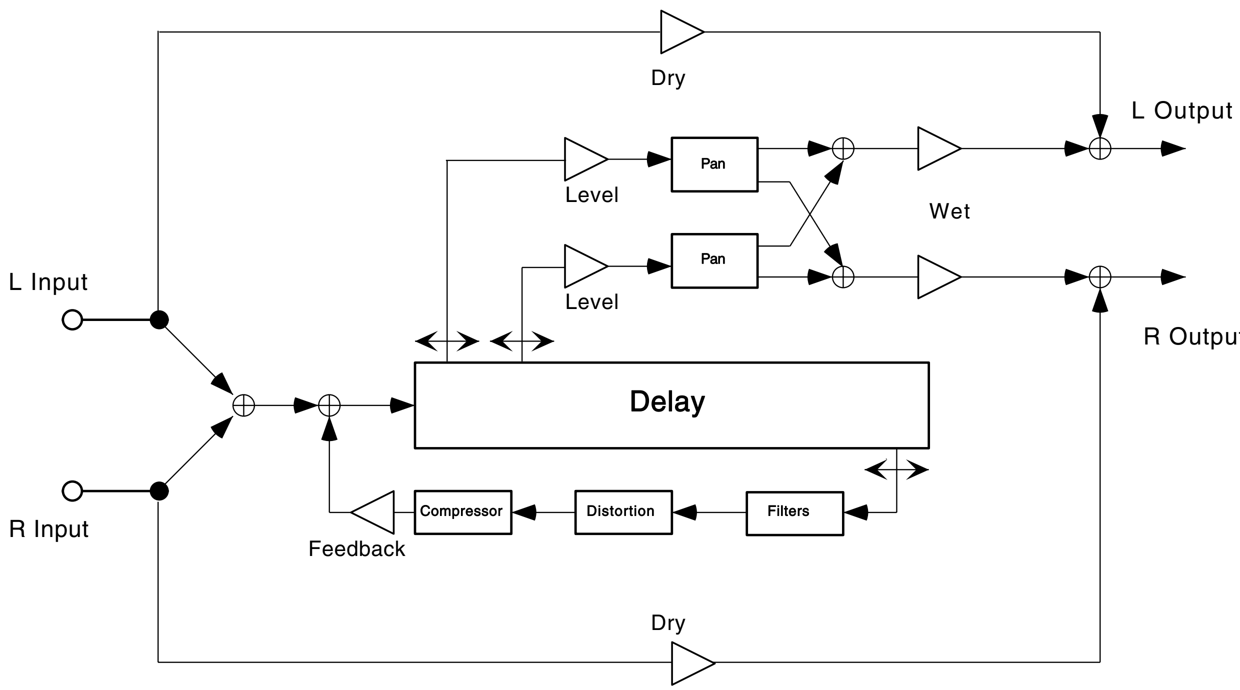


Figure 11 Degen Regen BPM

Parameters:

Page 1

Wet/Dry	-100 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Loop Gain	Off, -79.0 to 24.0 dB	Tempo	System, 1 to 255 BPM
Loop Lvl	-100 to 100%	Send Gain	Off, -79.0 to 24.0 dB
HF Damping	8 to 25088 Hz	LF Damping	8 to 25088 Hz

Page 2

LoopLength	0 to 32 bts	Mid1 Gain	-79.0 to 24.0 dB
LFO Period	1/24 to 32 bts	Mid1 Freq	8 to 25088 Hz
Bass Gain	-79.0 to 24.0 dB	Mid1 Width	0.010 to 5.000 oct
Bass Freq	8 to 25088 Hz	Mid2 Gain	-79.0 to 24.0 dB
Treb Gain	-79.0 to 24.0 dB	Mid2 Freq	8 to 25088 Hz
Treb Freq	8 to 25088 Hz	Mid2 Width	0.010 to 5.000 oct

Page 3

LpLFODepth	0.0 to 230.0 ct	Tap1 Delay	0 to 32 bts
LpLFOPhase	0.0 to 360.0 deg	Tap1 Level	0 to 100 %
T1LFODepth	0.0 to 230.0 ct	Tap1 Pan	-100 to 100%
T1LFOPhase	0.0 to 360.0 deg	Tap2 Delay	0 to 32 bts
T2LFODepth	0.0 to 230.0 ct	Tap2 Level	0 to 100 %
T2LFOPhase	0.0 to 360.0 deg	Tap2 Pan	-100 to 100%

Page 4

<table>
<tr><td>Comp Atk</td><td>0.0 to 228.0 ms</td><td>Comp Ratio</td><td>1.0:1 to 100.0:1, Inf:1</td></tr>
<tr><td>Comp Rel</td><td>0 to 3000 ms</td><td>Comp Thres</td><td>-79.0 to 0.0 dB</td></tr>
<tr><td>CompSmooth</td><td>0.0 to 228.0 ms</td><td>Dist Drive</td><td>0 to 96 dB</td></tr>
<tr><td></td><td></td><td>DistWarmth</td><td>8 to 25088 Hz</td></tr>
<tr><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>||||||||||||||||||||||||||||</td><td>Reduction</td></tr>
<tr><td></td><td>-dB 40 20 12</td><td>8 6 4 2 0</td><td></td></tr>
</table>

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.

Send Gain The input gain or amplitude to the **Degen Regen BPM** delay loop.

Loop Gain Controls the signal level of the signal which is fed back to the input of the delay line. If other elements of **Degen Regen BPM** were removed (set flat), then Loop Gain would cause the algorithm to become unstable above 0 dB. However other parameters interact resulting in a more complex gain structure. See also Loop Lvl.

Loop Lvl A convenience parameter which may be used to reduce the Fdbk Gain feedback strength. It may be helpful if you are used to dealing with feedback as a linear (percent) control. At **100**%, the feedback strength is as you have it set with Loop Gain. Lower levels reduce the feedback signal, so at **50**% the feedback signal is reduced by -6 dB from the selected Loop Gain level. Negative values polarity invert the feedback loop signal.

Tempo In **Degen Regen BPM**, Tempo is the 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. In this case, FXMods (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LF Damping The -3 dB frequency in Hz of a one-pole highpass filter (6 dB/octave) placed in the feedback path of the delay line. The signal does not go through the filter the first time through the delay line. Multiple passes through the feedback will cause the signal to become more and more bright (removing low frequencies).

HF Damping The -3 dB frequency in Hz of a one-pole lowpass filter (-6 dB/octave) placed in the feedback path of the delay line. The signal does not go through the filter the first time through the delay line. Multiple passes through the feedback will cause the signal to become more and more dull.

LoopLength The delay length of the feedback tap. If feedback is turned up from **0%**, this parameter sets the repeating delay loop length. In **Degen Regen BPM**, the loop length is specified as a fraction or multiple of the tempo, in "beats." The length of a delay loop in seconds can be calculated from beats as T = (beats/tempo) ∗ 60.

LFO Period The feedback tap and the output taps lengths can be modulated with an LFO internal to the effects processor. The rate at which the tap positions move are tied to a common period control (time for one complete cycle) which is expressed in beats. The LFO Period control is specific to **Degen Regen BPM**. The depth of modulation is specified by the LpLFODepth parameter. Frequency in Hz can be calculated from the period in beats as F = tempo/(beats ∗ 60). Since this moving delay tap is part of the feedback path through the delay, subsequent passes of the signal through the delay may result in some strange pitch modulations. It is possible to set LFO Period with LoopLength so that alternate passes through the loop detune then retune the signal (for example, set the LFO period to double the LoopLength). The maximum pitch shift up is not identical to the maximum pitch shift down, so the alternating detune/retune effect is not perfect.

Bass Gain The amount of boost or cut in dB that the bass shelving filter should apply to the low frequency signal components. 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. Since the filters are in the delay feedback loop, the cut or boost is cumulative on each pass the sound makes through the loop.

Bass Freq The center frequency of the bass shelving filter in intervals of one semitone.

Treb Gain The amount of boost or cut in dB that the treble shelving filter should apply to the high frequency signal components. 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. Since the filters are in the delay feedback loop, the cut or boost is cumulative on each pass the sound makes through the loop.

Treb Freq The center frequency of the treble shelving filter in intervals of one semitone.

Midn Gain The amount of boost or cut in dB that the parametric filter should apply to the specified signal 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. Since the filters are in the delay feedback loop, the cut or boost is cumulative on each pass the sound makes through the loop.

Midn Freq The center frequency of the parametric EQ in intervals of one semitone. The boost or cut will be a maximum at this frequency.

Midn Width The bandwidth of the parametric EQ may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow (high-Q) filter response. Large values result in a very broad response.

LpLFODepth The feedback (loop) delay tap will have its position modulated by an LFO (internal to the FX processor) if the LpLFODepth parameter is non-zero. A moving tap on a delay line will result in a pitch shift, and LpLFODepth sets the maximum pitch shift (up and down) in cents.

LpLFOPhase Specifies the phase angle of the feedback (loop) LFO relative to the output tap LFOs and the system (or MIDI) 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 (or MIDI) tempo clock is turned on , the LFOs are synchronized to the clock with absolute phase.

T*n*LFODepth The output delay taps (1 and 2) will have their positions modulated by an LFO (internal to the FX processor) if the TnLFODepth parameter is non-zero. A moving tap on a delay line will result in a pitch shift, and TnLFODepth sets the maximum pitch shift (up and down) in cents.

T*n*LFOPhase Specifies the phase angle of the output LFO tap (1 or 2) relative to the other output LFO tap, the feedback (loop) LFO tap, and the system (or MIDI) 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 (or MIDI) tempo clock is turned on, the LFOs are synchronized to the clock with absolute phase.

Tap*n* Delay The delay length of the output tap 1 or 2. In **Degen Regen BPM**, the tap length is specified as a fraction or multiple of the tempo, in "beats." The length of a delay tap in seconds can be calculated from beats as T = (beats/tempo) ∗ 60.

Tap*n* Level The level of the output tap 1 or 2 expressed as a percent.

Tap*n* Pan The output taps 1 and 2 are mono sources that can be panned to the left or right output channels. A pan setting of **-100%** is fully left while **100%** is fully right.

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.

CompSmooth A lowpass filter in the compressor 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.

Comp Ratio The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

Comp Thres The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

Dist Drive Applies a boost to the feedback 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 feedback amount or turn on the compressor as the drive is increased.

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.

139 Switch Loops

Looped delay lines with input switching

PAUs: 2

Switch Loops allows you to run up to four parallel recirculating delay lines of different lengths, switching which delay line(s) are receiving the input signal at a given moment. The stereo input is summed to mono and sent to any of the four delay lines. You can select which delay lines are receiving input with the DlySelect parameters.

The gain in decibels of each of the four delays can be set individually. The amount of feedback to apply to each delay is set with a DecayRate parameter. The DecayRate controls how many decibels the signal will be reduced for every second the signal is recirculating in the delay.

The length of the delays are set based on tempo (system tempo or set locally) and duration in beats. Assuming a 4/4 time signature with tempo beats on the quarter note, 8/24 bts is an eighth triplet (8/24 equals 1/3 of a quarter note), 12/24 bts is an eighth, 16/24 bts is a quarter triplet, and 1 bts is a quarter note duration. Dividing the quarter note into 24ths, allows delay lengths based on the most common note lengths. To determine a delay length in seconds, divide the length of the delay (in beats) by the tempo and multiply by 60 seconds/minute (beats/tempo ∗ 60).

Switch Loops has a few more specialized parameters. HF Damping controls a one pole lowpass filter on each of the delay lines. Max Fdbk overrides all of the DecayRate parameters and prevents the signals in the delay lines from decaying at all. Fdbk Kill will override the DecayRate parameters and the Max Fdbk parameter by completely turning of the feedback for all the delays. Fdbk Kill stops all the delay line recirculation.

The outputs of all the delay lines are summed, and the output gain is applied to the mono result which can be panned between the two output channels.

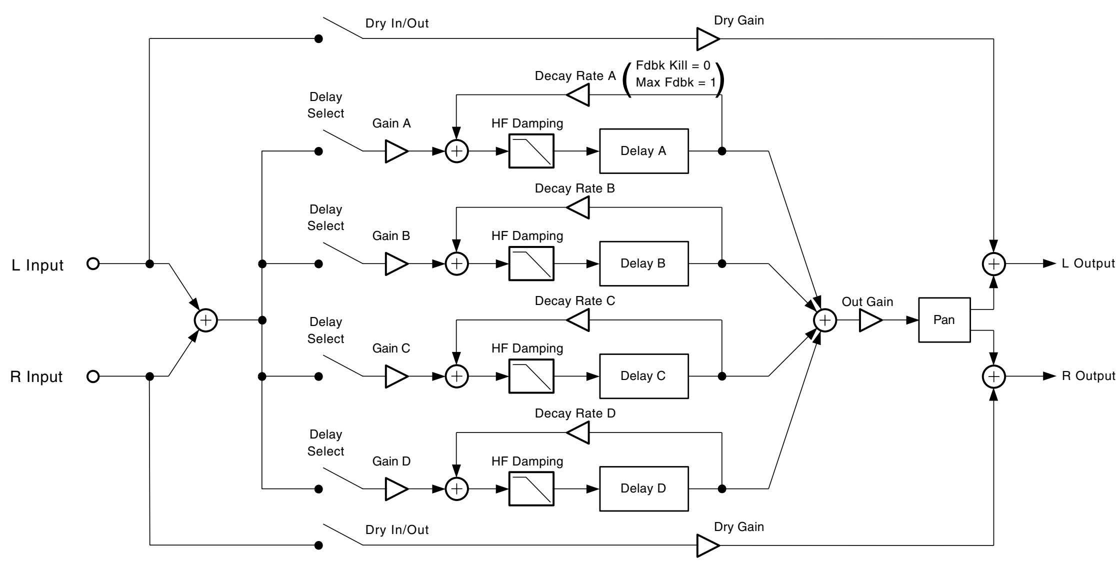


Figure 12 **Switch Loops**

Parameters:

Page 1

Dry In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Dry Gain	Off, -79.0 to 24.0 dB	Tempo	System, 1 to 255 BPM
Fdbk Kill	On or Off	Pan	-100 to 100 %
Max Fdbk	On or Off	HF Damping	8 to 25088 Hz

Page 2

DlySelect1	Off, A, B, C, D		
DlySelect2	Off, A, B, C, D		
DlySelect3	Off, A, B, C, D		
DlySelect4	Off, A, B, C, D		

Page 3

Dly Len A	0 to 32 bts	Dly Len C	0 to 32 bts
DecayRateA	0.0 to 230.0 dB/s	DecayRateC	0.0 to 230.0 dB/s
Gain A	Off, -79.0 to 24.0 dB	Gain C	Off, -79.0 to 24.0 dB
Dly Len B	0 to 32 bts	Dly Len D	0 to 32 bts
DecayRateB	0.0 to 230.0 dB/s	DecayRateD	0.0 to 230.0 dB/s
Gain B	Off, -79.0 to 24.0 dB	Gain D	Off, -79.0 to 24.0 dB

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

Dry In/Out If set to **In**, Dry In/Out allows the dry input signal to be added to the final algorithm output.

Dry Gain If Dry In/Out is **In**, then Dry Gain controls the level of the dry input signal that is summed to the final algorithm output.

Fdbk Kill Forces the delay recirculation of all delay lines to stop by turning off the delay line feedback. Fdbk Kill provides a quick way to silence the algorithm to start over with new sounds in the delays. Fdbk Kill overrides the Max Fdbk and DecayRate parameters.

Max Fdbk Prevents the recirculating delay lines from decaying by turning the delay line feedback fully on. Max Fdbk overrides the DecayRate parameters, but does not function when Fdbk Kill is **On**.

Tempo Tempo is the 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. In this case, FXMods (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

Pan The summed mono signal from the delay lines may be panned between left and right output channels. **-100**% is panned fully left, **0**% is centered, and **100**% is fully right.

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

DlySelect*n* You select which delay lines (A, B, C, or D) receive the mono input signal with the DlySelect (1, 2, 3, or 4) parameters. Since there are four delay lines, you can turn on none, 1, 2, 3, or 4 of the delay lines. All four of the DlySelect parameters are equivalent—it doesn't matter which you use. If you turn on a particular delay line in more than one DlySelect parameter, it's the same as turning it on in just one DlySelect parameter.

Dly Len *n* The delay length of the delay line n (n = A, B, C, or D). If the DecayRate for the delay is low or Max Fdbk is **On**, this parameter sets the repeating delay loop length for this delay. The delay length is specified as a fraction or multiple of the tempo, in "beats." The length of a delay loop in seconds can be calculated from beats as $T = (\text{beats}/\text{tempo}) * 60$.

DecayRate*n* The rate at which the delay line n (n = A, B, C, or D) will decay or reduce in level. DecayRate controls a feedback level which is calculated based on DecayRate and Dly Len. By basing the feedback gain on DecayRate, all four of the delay lines can decay at the same rate in spite of differing delay lengths. DecayRate is expressed as decibels of signal reduction per second.

Gain *n* The level of the delay n (n = A, B, C, or D) output tap expressed in decibels.

140 Moving Delay

Generic stereo moving delay lines

PAUs: 1

Moving Delay is identical to **Dual MovDelay** except that the algorithm now has stereo controls rather than dual mono. This means all the controls except L Pan and R Pan are no longer dual left and right but are ganged into single controls controlling both left and right channels.

Parameters:

Page 1

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

Page 2

Delay	0.0 to 1000.0 ms		
LFO Mode	ChorTri, ChorTrap, Delay, Flange		
LFO Rate	0.00 to 10.00 Hz		
LFO Depth	0.0 to 200.0 %		
Feedback	-100 to 100 %		
HF Damping	8 to 25088 Hz		

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

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

L Pan, R Pan The output panning position of each moving delay circuit. **0**% is center; Negative values pan left, while positive values pan right.

Delay Adjusts the delay time for the moving delay circuits, which is the center of LFO excursion.

LFO Mode Adjusts the LFO excursion type. In Flange mode, the LFO is optimized for flange effects and LFO Dpth adjusts the excursion amount. In ChorTri and ChorTrap modes, the LFO is optimized for triangle and trapezoidal pitch envelopes respectively, and LFO Dpth adjusts the amount of chorus detuning. In Delay mode, the LFO is turned off leaving a basic delay. LFO Rate and LFO Dpth in Delay mode are disabled.

LFO Rate Adjusts the LFO speed for the moving delay circuits.

LFO Depth In Flange LFO mode, this adjusts an arbitrary LFO excursion amount. In ChorTri and ChorTrap modes, this controls the chorus detune amount. In delay mode, this is disabled.

Feedback Adjusts the level of the moving delay circuits' output signal fed back into their own inputs. Negative values polarity invert the feedback signal.

HF Damping Adjusts the cutoff frequency of a 1-pole (6dB/oct) lowpass filter in the moving delay circuits.

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.

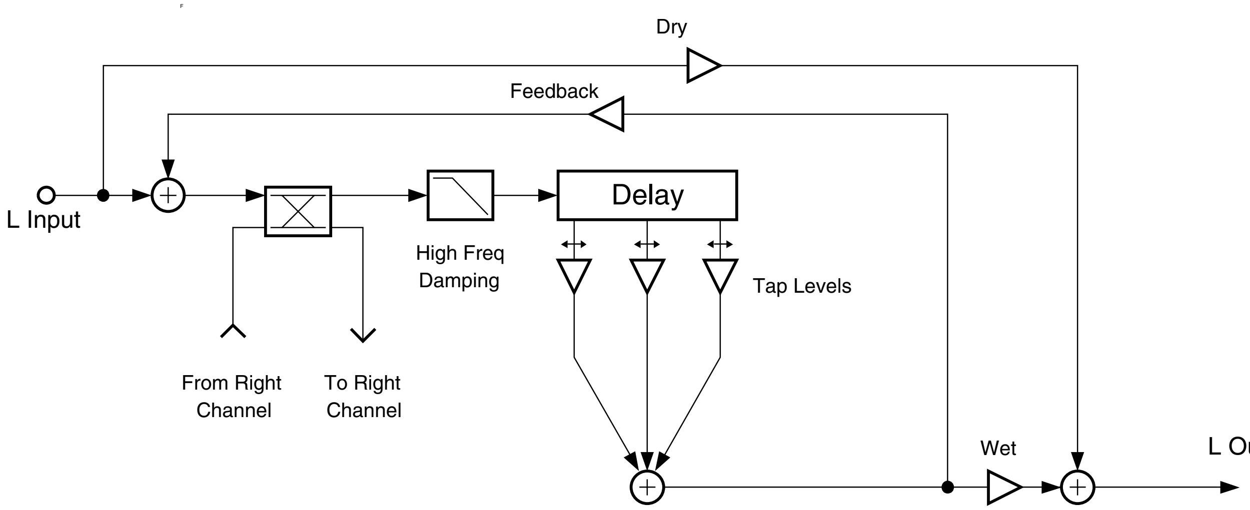


Figure 10-13 Block diagram of left channel of Chorus 2

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 10-13 is a simplified block diagram of the left channel of Chorus 2.

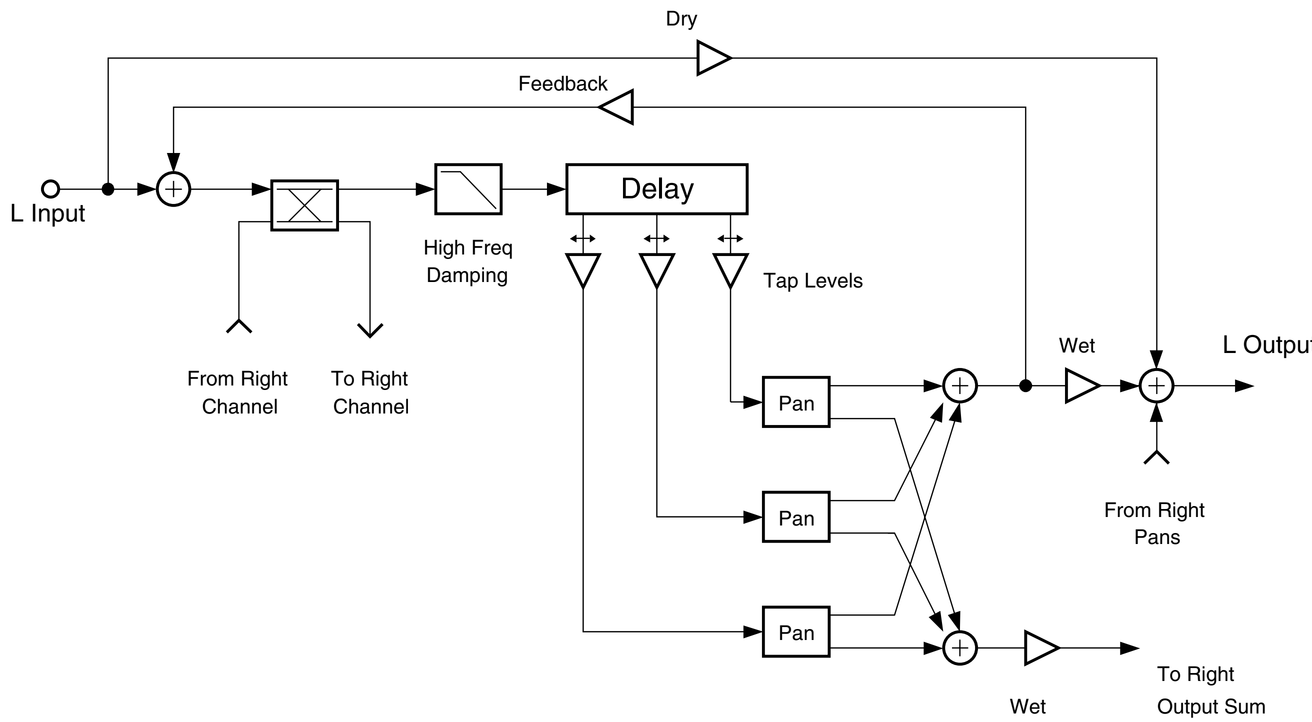


Figure 10-14 Block Diagram of Left Channel of Dual Chorus 2 (right channel is similar)

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

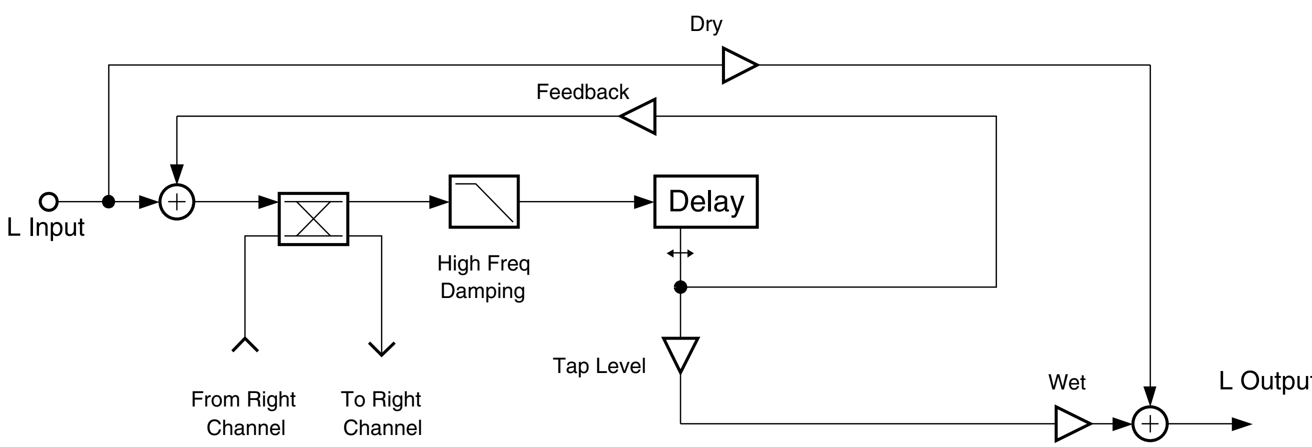


Figure 10-15 Block diagram of left channel of Chorus 1 (right channel is the same)

Chorus 1 uses just 1 unit allocation and has one delay tap. Figure 10-15 is a simplified block diagram of the left channel of Chorus 1.

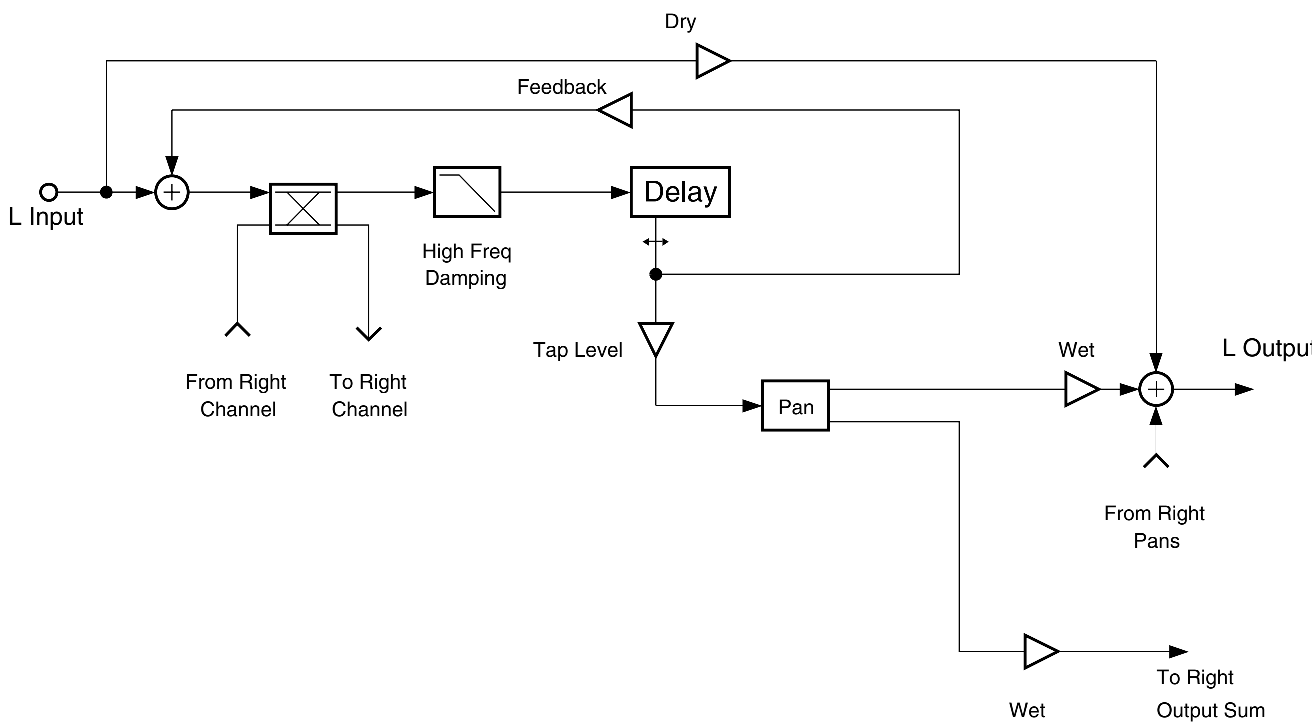


Figure 10-16 Block diagram of left channel of Dual Chorus 1 (right channel is similar)

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.

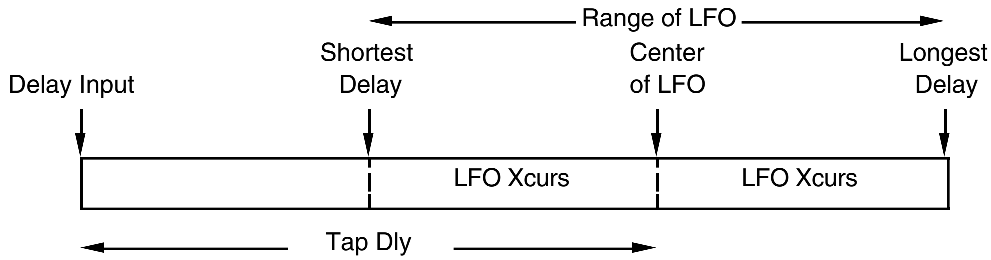


Figure 10-17 Delay for a Single LFO

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.

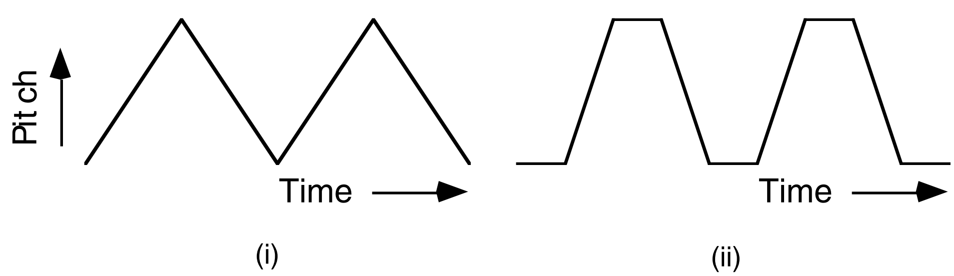


Figure 10-18 Pitch Envelopes (i) Triangle and (ii) Trapzoid

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.

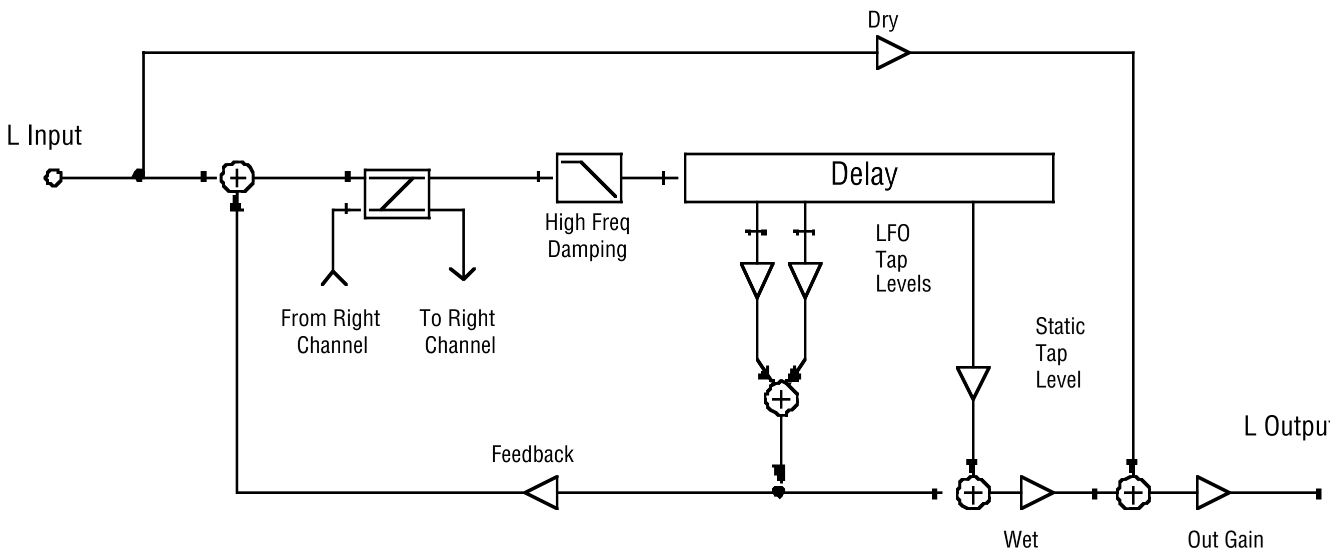


Figure 10-19 Simplified block diagram of the left channel of Flanger 1 (right channel is similar)

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

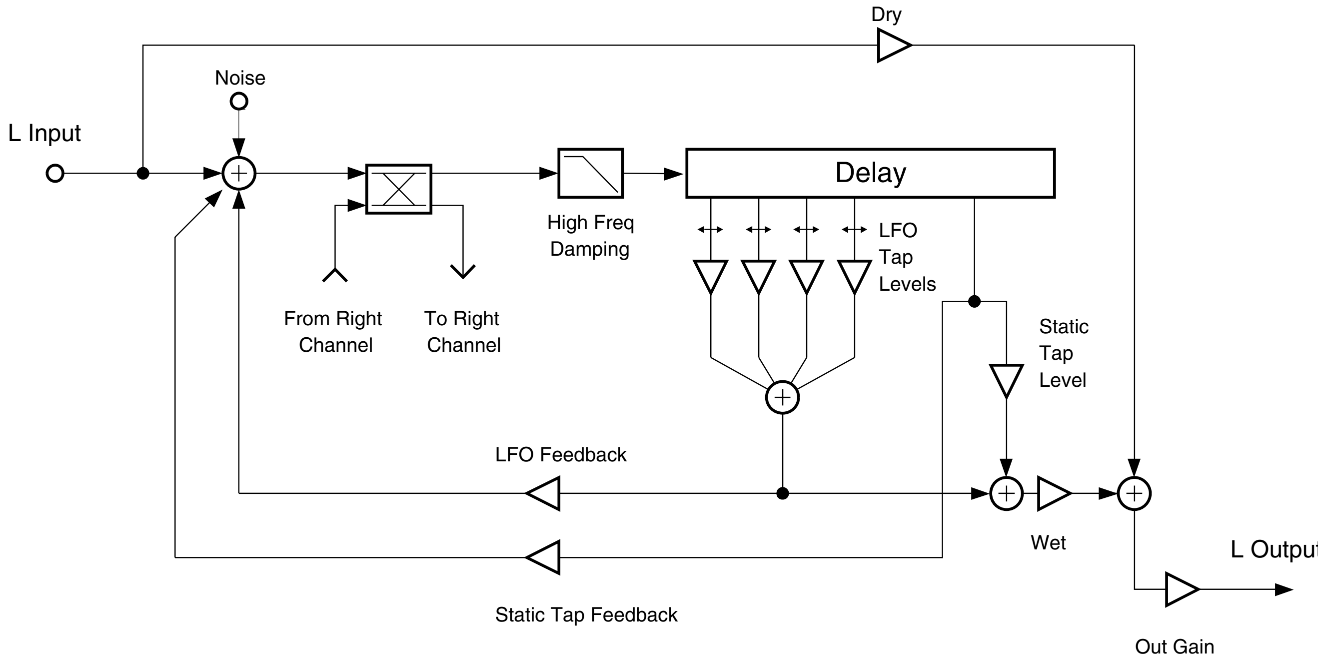


Figure 10-20 Simplified block diagram of the left channel of Flanger 2 (right channel is similar)

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 10-20. 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.

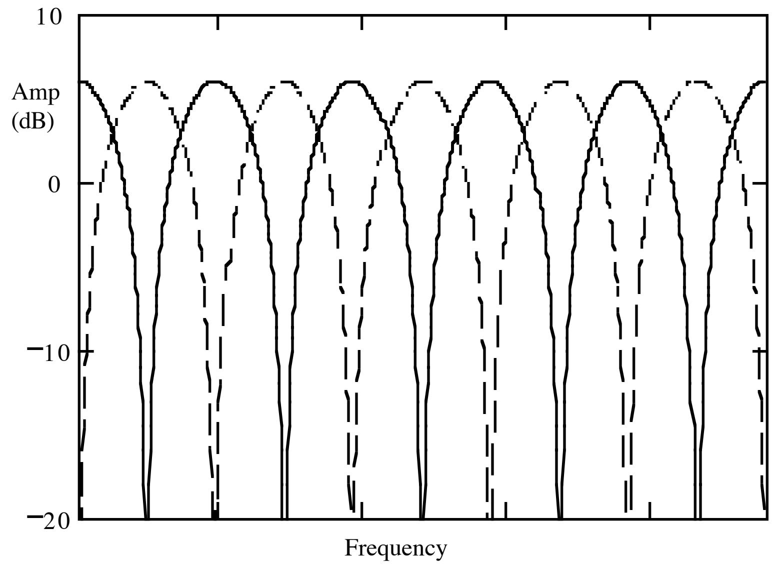


Figure 10-21 Comb Filters : Solid Line for Addition; Dashed Line for Subtraction

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.

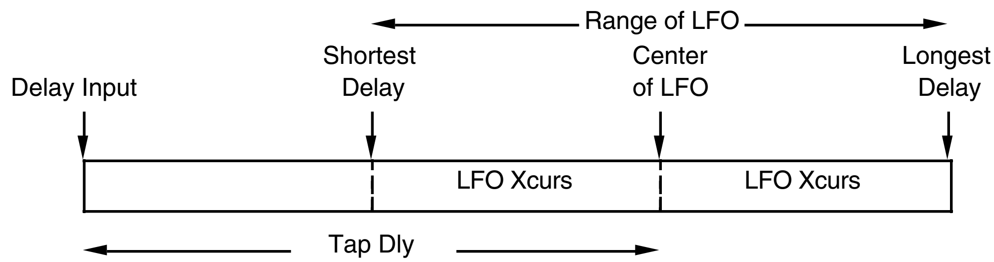


Figure 10-22 Delay for a Single LFO

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 K2661, 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 K2661, 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.

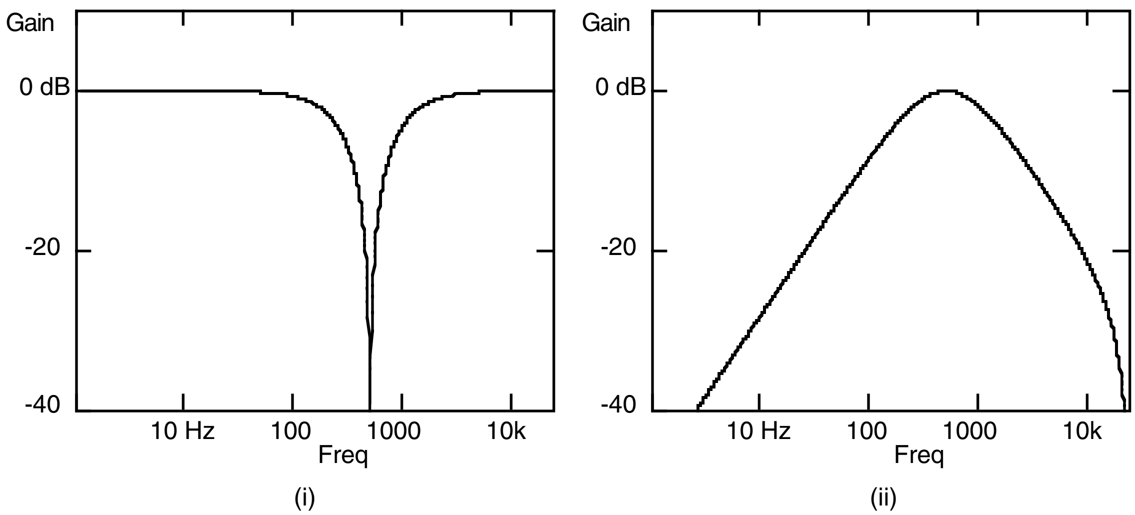


Figure 10-23 Response of typical phaser with (i) Wet/Dry = 50% and (ii) WetDry = -50%.

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.

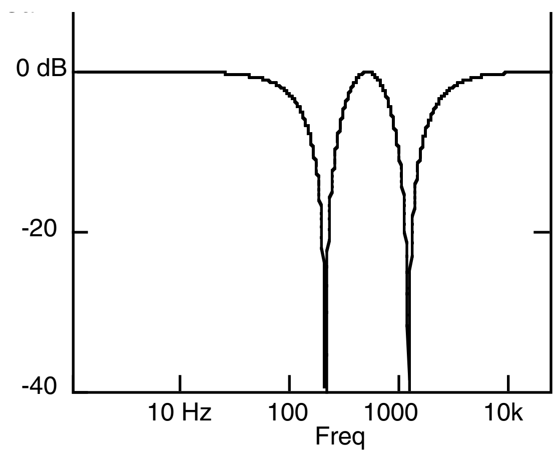


Figure 10-24 Response of LFO Phaser Twin with Wet/Dry set to 100%.

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.

161 Allpass Phaser 3

Allpass filter phasers

PAUs: 3

The allpass phasers are algorithms that use allpass filters to achieve a phaser effect. These algorithms do not have built in LFOs, so like **Manual Phaser**, any motion must be supplied with an FXMod. Unlike the other phasers, the allpass phasers use high order allpass filters. The order of the allpass filters sets the number of notches that will appear in the frequency response when the dry and filtered signals are mixed. The number of notches in the frequency response ranges from 3 to 6 for **Allpass Phaser 3** . The allpass phaser algorithms use a typical signal routing with wet/dry and cross-coupled feedback. A different number of notches may be chosen for the feedback path than for the direct output.

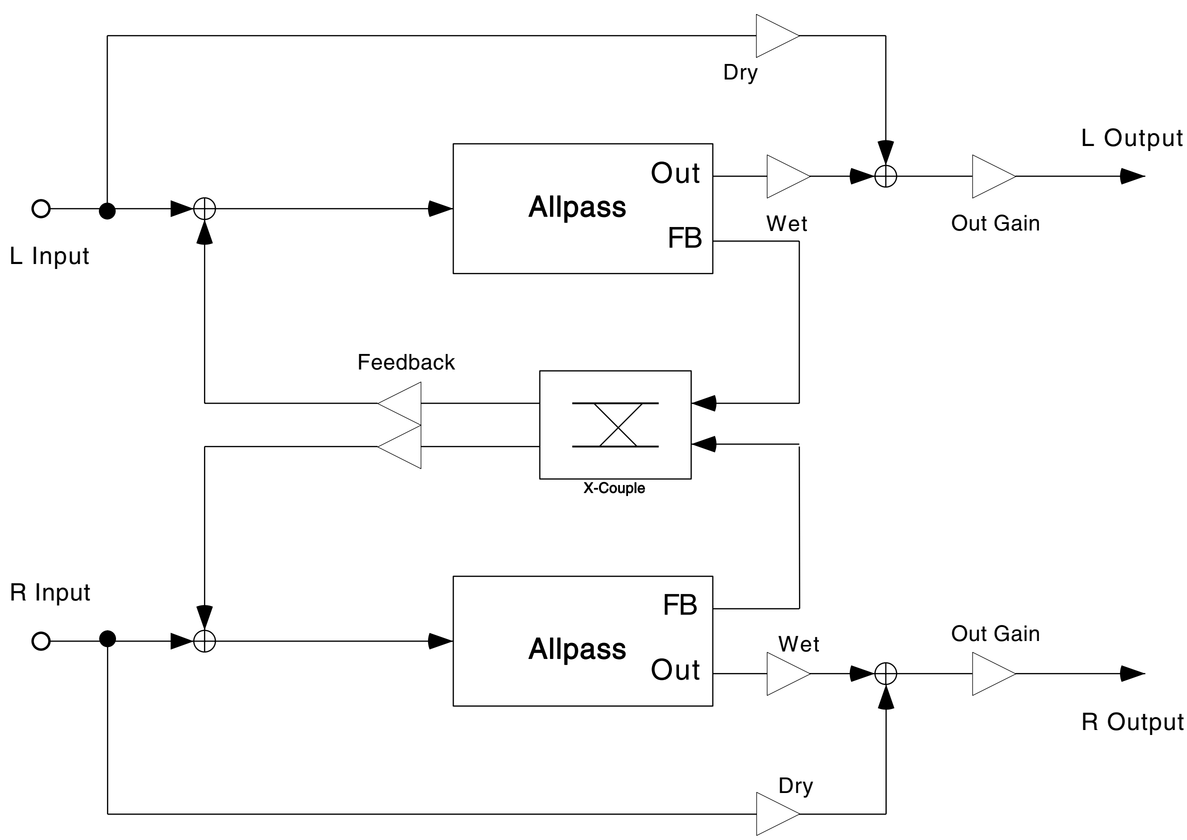


Figure 25 An allpass phaser

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.

As mentioned earlier, allpass phasers leave the phaser motion up to you, so they have no built in LFOs. 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.

When feedback is used, it can greatly exaggerate the peaks and notches, producing a much more resonant sound. Cross-coupling (XCouple) the feedback between the left an right channels increases the complexity of the frequency response.

In the figure above, you’ll notice that the spacing of the notches and peaks are not harmonically related. When a lot of feedback is used, the non-harmonic structure produces very bell-like tones, particularly with XCouple set to **100%.** (Don’t modulate the frequencies to get this effect.) Try experiments using different allpass orders for the feedback, different frequency arrangements, changing the sign (+/-) of the feedback (Fdbk Level) parameter, and different input sources (drums are a good starting point).

Parameters:

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%
LCenterFreq	8 to 25088 Hz	RCenterFreq	8 to 25088 Hz
FB APNotch	3 to 6 or 7 to 10	OutAPNotch	3 to 6 or 7 to 10

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.

XCouple Determines how much of the right feedback signal to feed into the left input channel and how much left feedback to feed into the right input channel. When increasing cross-coupling, the amount of feedback from one channel into its own input is reduced, so that at **100%** the left feeds back entirely to the right channel and vice versa. [Stereo versions only]

CenterFreq The nominal center frequency of the phaser filter. The frequency LFO modulates the phaser filter centered at this frequency. There are separate left and right controls in the stereo version.

FB APNotch The number of notches the allpass filter can produce when summed with a dry signal. Used in the feedback loop. Higher values produce more resonant peaks, for a more complex resonant structure.

OutAPNotch The number of notches the allpass filter can produce when summed with a dry signal. Used on the algorithm output. Higher values produce a steeper, longer phase response resulting in more peaks and notches when combined with the dry signal.

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 10-26 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.

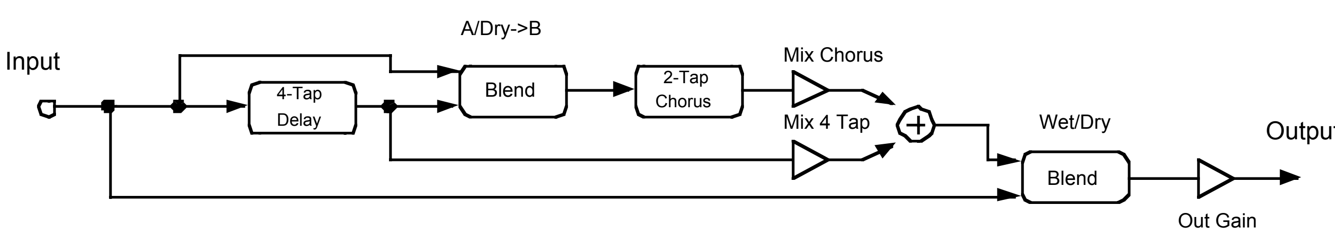


Figure 10-26 An example of routing using Chorus+4Tap

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 what 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 what 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

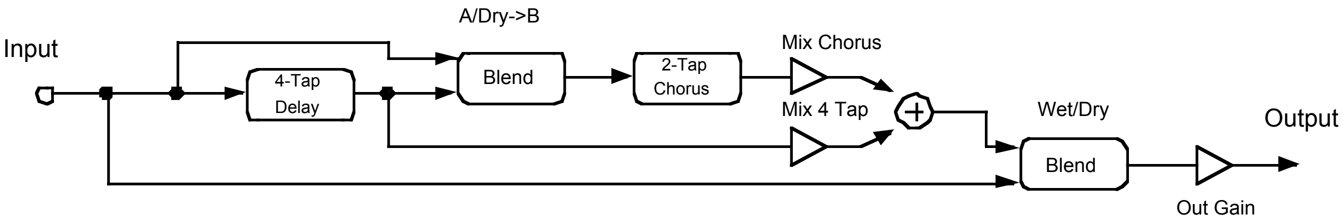
A family of combination effect algorithms

PAUs: 2

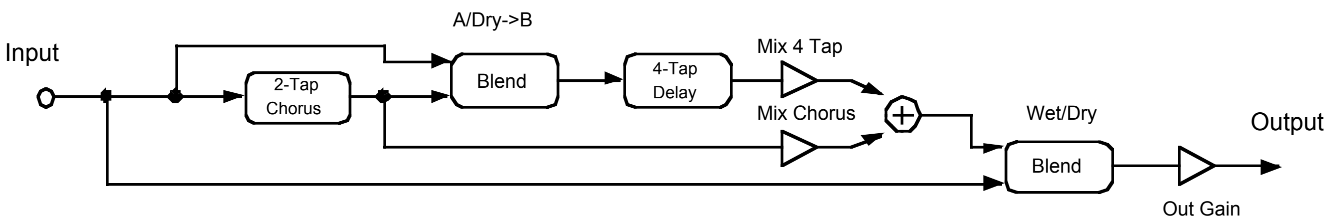
Signal Routing

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 10-27.

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 10-27 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.

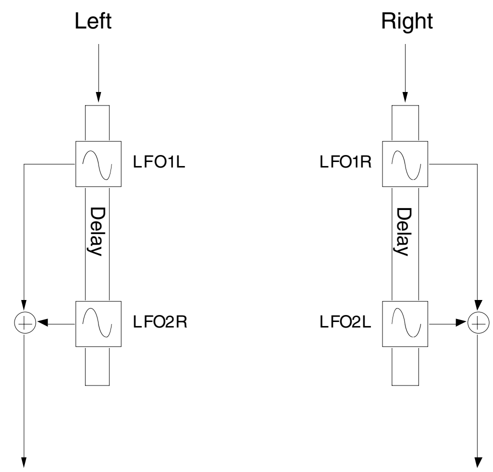


Figure 10-28 LFO delay taps in the configurable chorus and flange

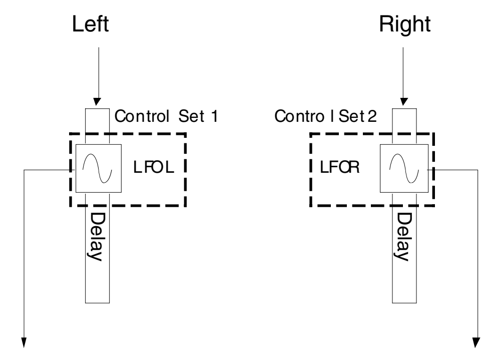


Figure 10-29 LFO control in Dual1Tap mode

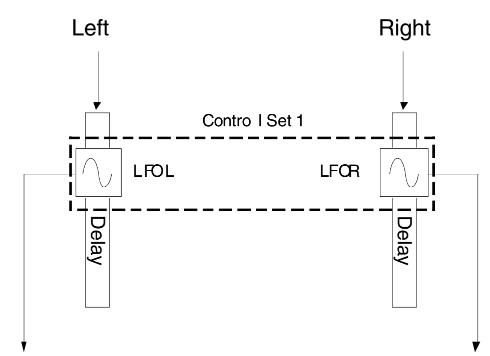


Figure 10-30 LFO control in Link1Tap mode

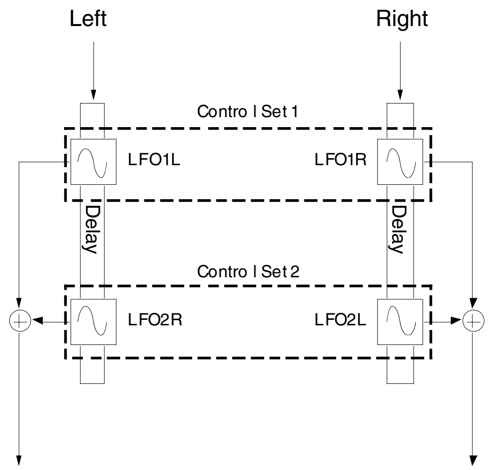


Figure 10-31 LFO control in Link2Tap mode

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.

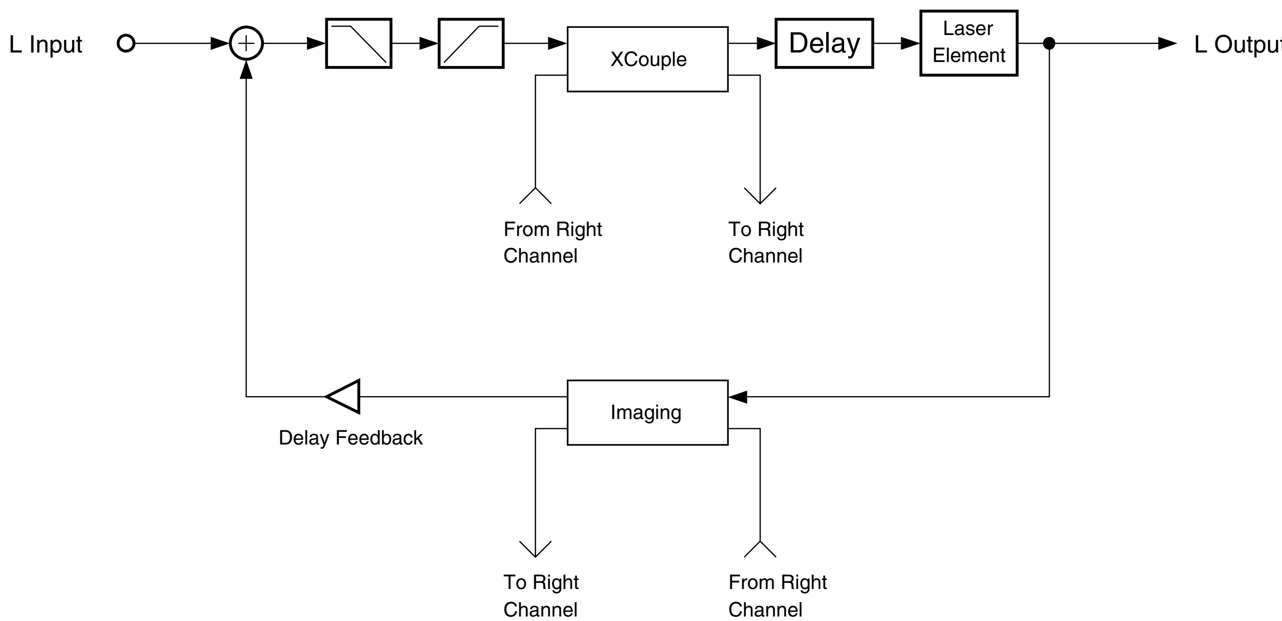


Figure 10-32 Laser Delay (left channel)

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 K2661'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.

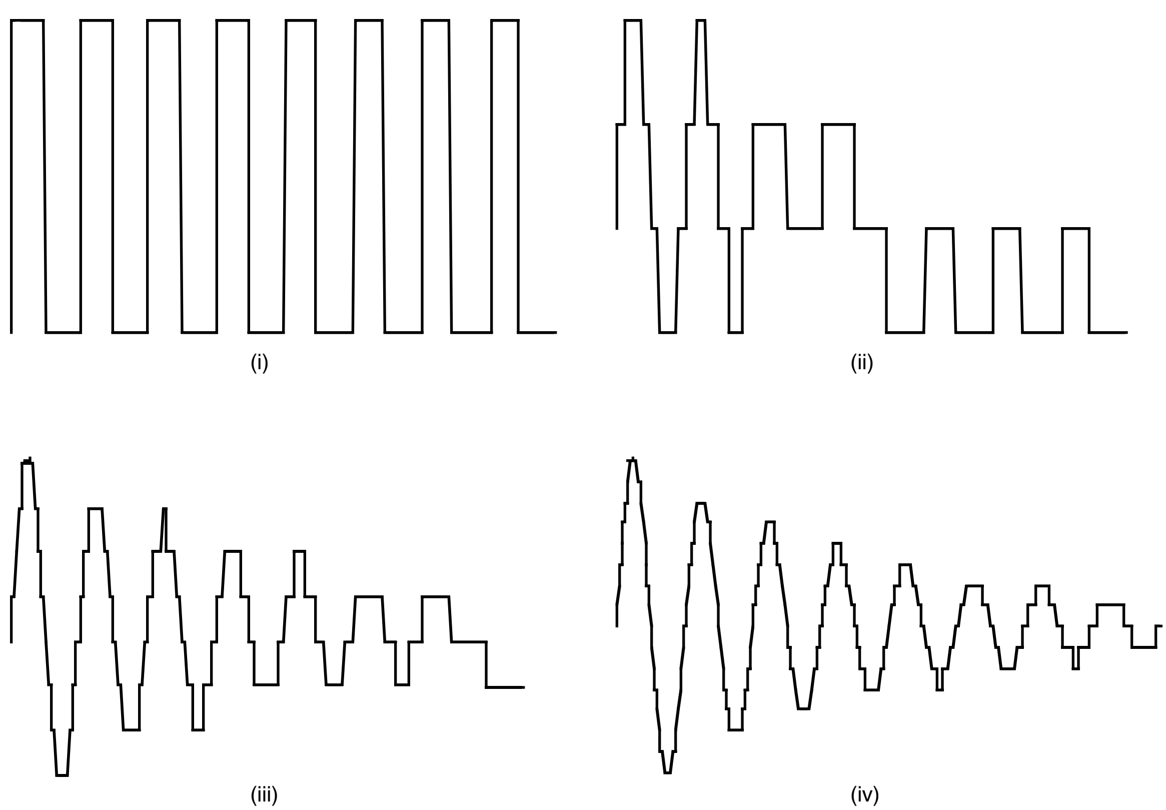


Figure 10-33 A decaying sine wave represented with different word lengths: (i) 1-bit, (ii) 2-bit, (iii) 3-bit, (iv) 4-bit.

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.

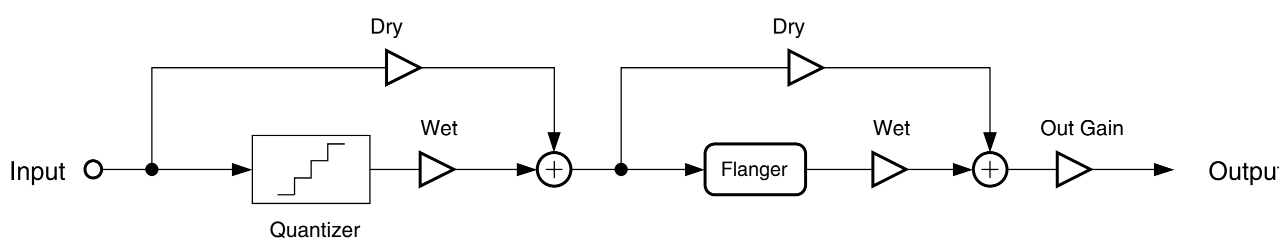


Figure 10-34 Block diagram of one channel of Quantize+Flange.

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.

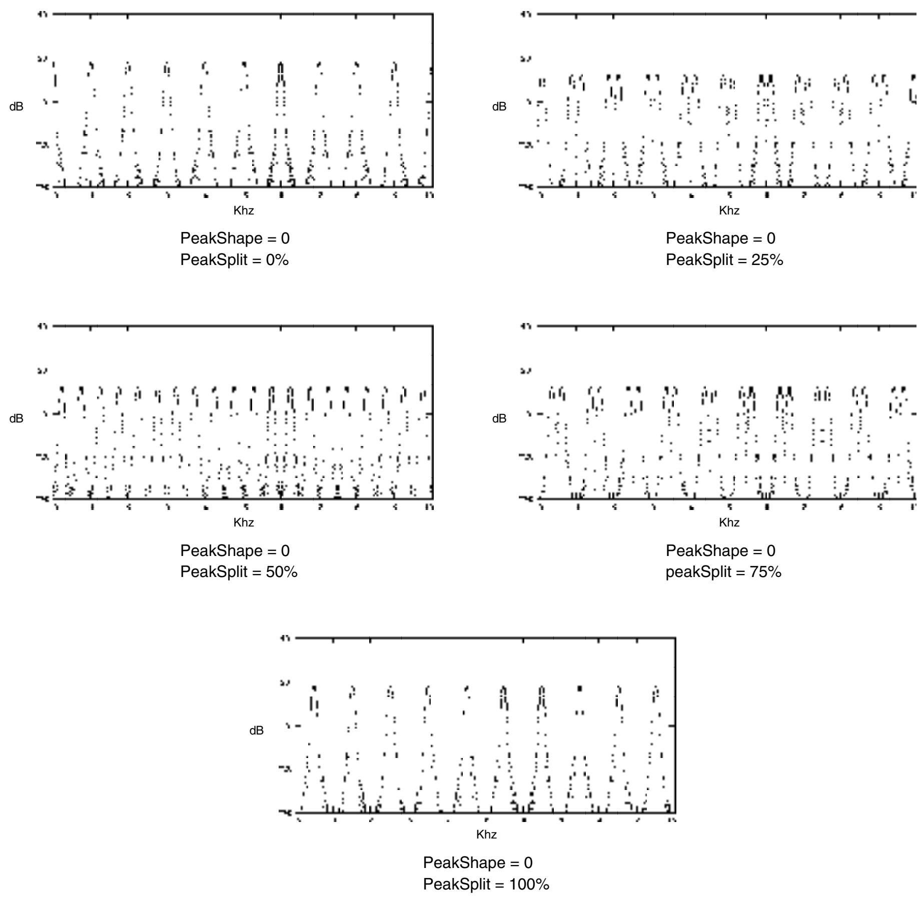
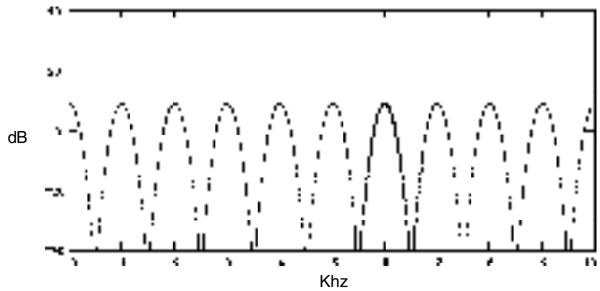


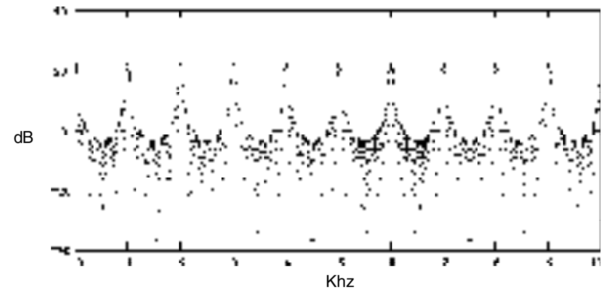
Figure 10-35 Response of Pitcher with different PkSplit settings. Pitch is C6 and PkShape is 0.

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 10-36 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

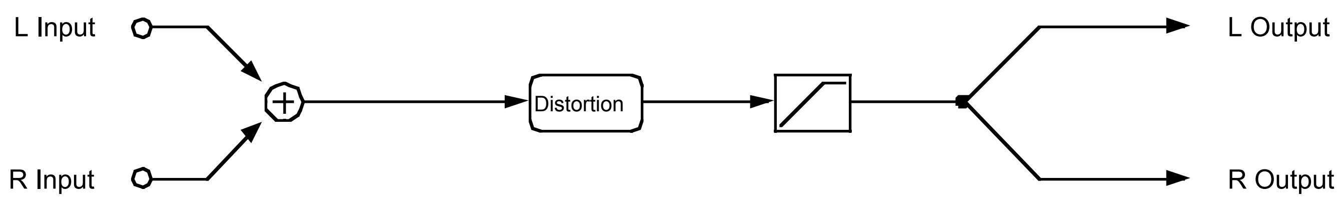


Figure 10-37 Block diagram of Mono Distortion

Mono Distortion sums its stereo input to mono, performs distortion followed by a highpass filter and sends the result as centered stereo.

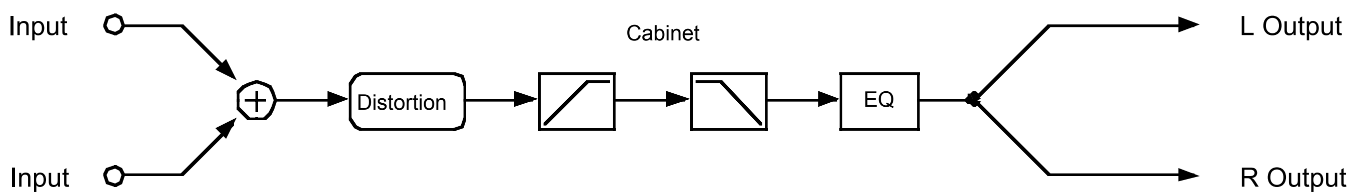


Figure 10-38 Block diagram of MonoDistort + EQ

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.

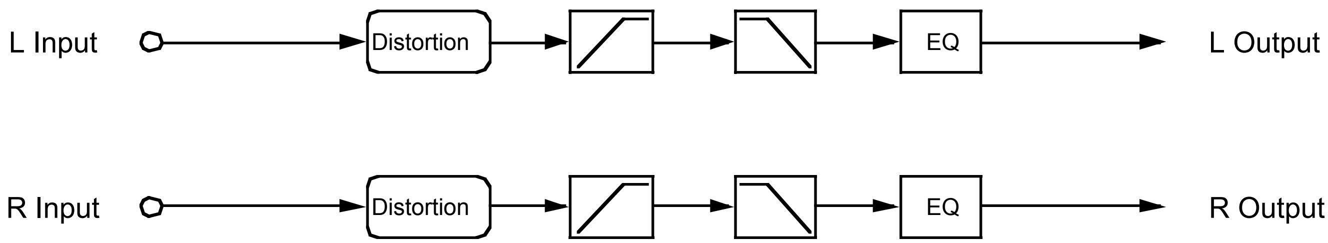


Figure 10-39 Block diagram of StereoDistort+EQ

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.

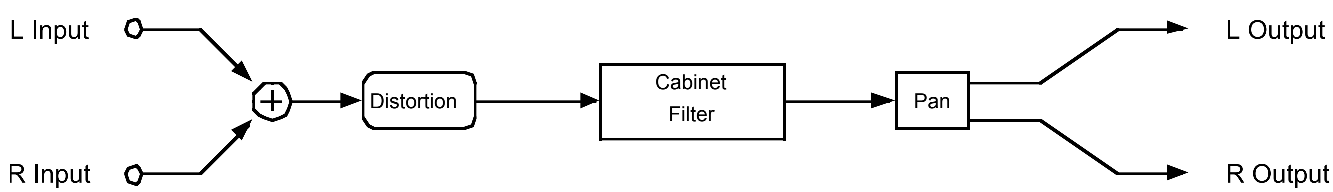


Figure 10-40 Block diagram of MonoDistort + Cab

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.

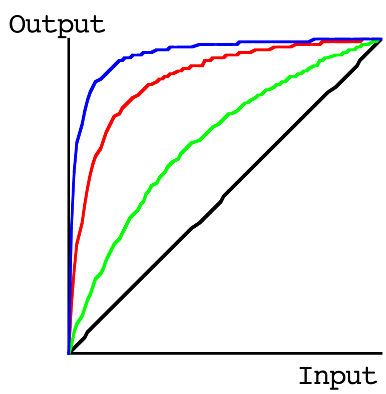


Figure 10-41 Input/Output Transfer Characteristic of Soft Clipping at Various Drive Settings

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.

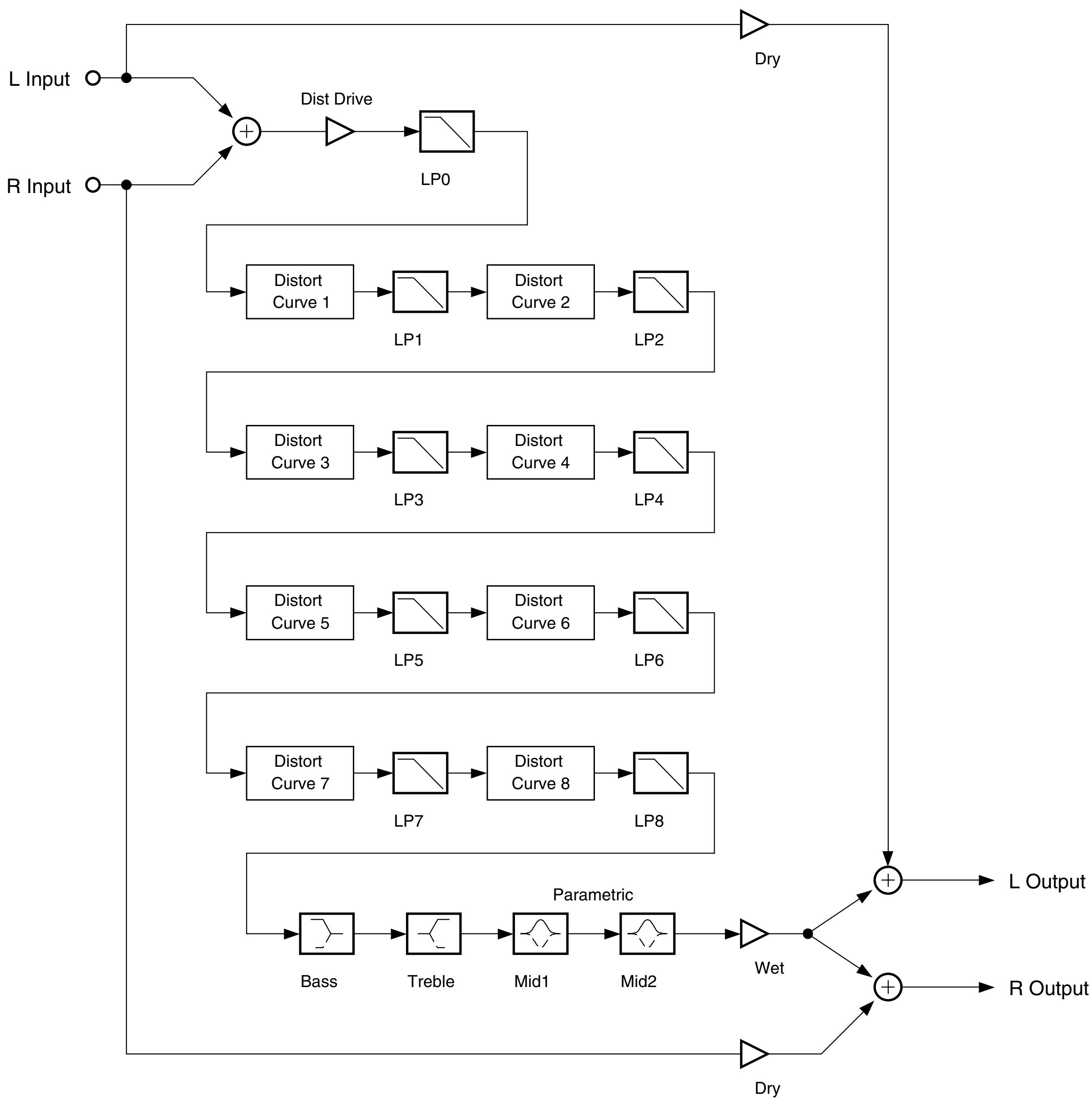


Figure 10-42 Block diagram of PolyDistort + EQ

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%

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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.

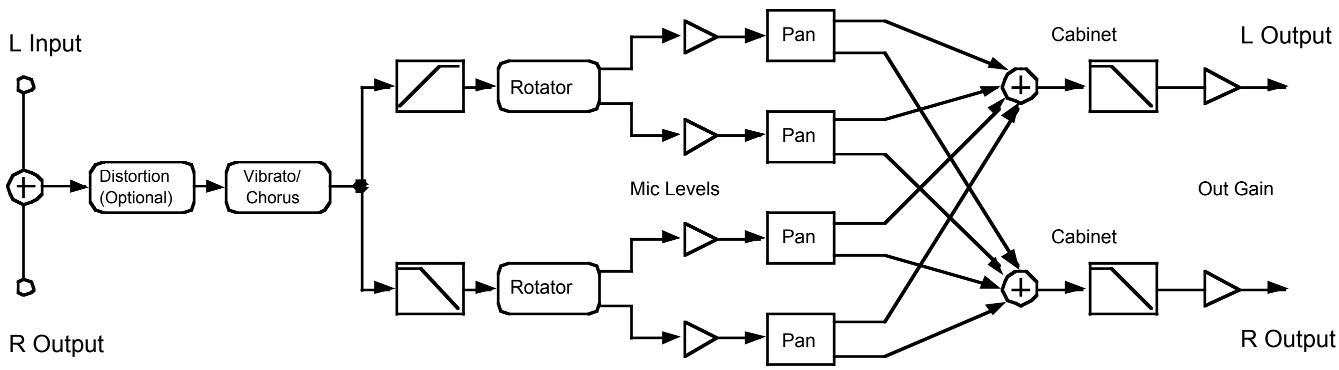

Figure 10-43 Block diagram of VibChor+Rotor

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.

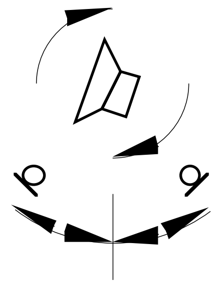

Figure 10-44 Rotating speaker with virtual microphones

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 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.

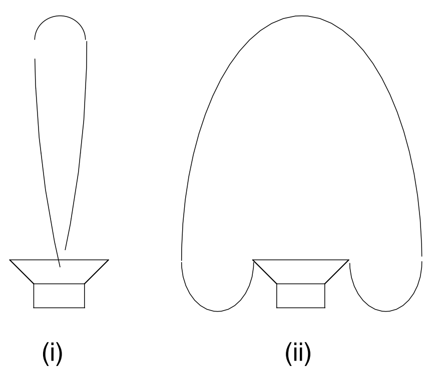

Figure 10-45 Acoustic beams for (i) low frequency driver and (ii) high frequency driver

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

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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.

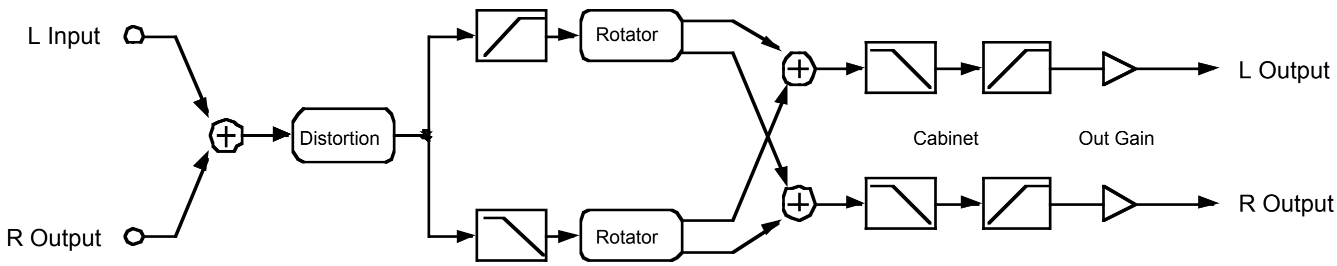


Figure 10-46 Block diagram of Distort + Rotary

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.

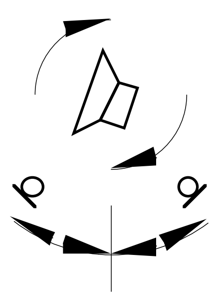


Figure 10-47 Rotating speaker with virtual microphones

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**.

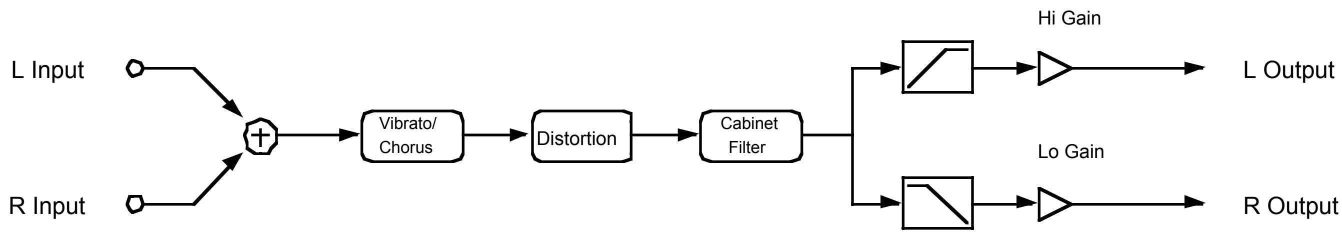


Figure 10-48 Block diagram of KB3 FXBus

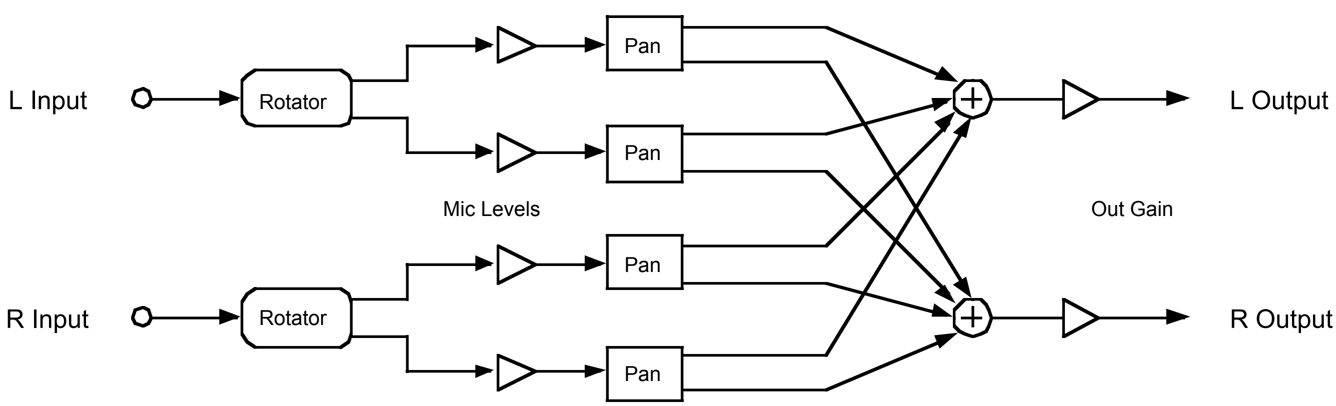


Figure 10-49 Block diagram of KB3 AuxFX

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.

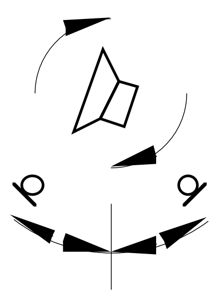


Figure 10-50 Rotating speaker with virtual microphones

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.

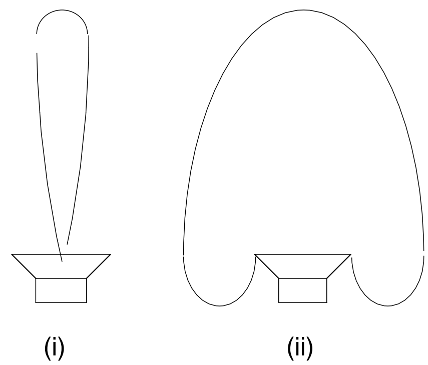


Figure 10-51 Acoustic beams for (i) low frequency driver and (ii) high frequency driver

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.

Rotary Effects

738 VC+Dist+1Rotor 2
739 VC+Dist+HiLoRotr
740 VC+Tube+Rotor 4
741 Rotor 1
742 VC+Dist+HiLoRot2

Rotating speaker algorithms

PAUs: 1 for **Rotor 1**
2 each for **VC+Dist+1Rotor 2**, **VC+Dist+HiLoRotr**, and **VC+Dist+HiLoRot2**
4 for **VC+Tube+Rotor 4**

The rotary algorithms contain multiple effects designed for the Hammond B3® emulation (KB3 mode). These effects may include the Hammond® vibrato/chorus, amplifier distortion, cabinet emulation and rotating speaker (Leslie®). A variety of rotating speaker algorithms have been designed to deal with different circumstances Some of the algorithms are designed to trade off features or model quality to allow the rotating speaker model to work in fewer PAUs.

The first effect in the chain is often 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 **VC+Tube+Rotor 4**, the vibrato chorus has been carefully modeled after the electromechanical vibrato/chorus in the B3. The vibrato/chorus in the other smaller algorithms use 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 fully modeled vibrato/chorus.

The final section of each of the rotary algorithms is the rotating speaker routine. The various algorithms may trade off some features of the rotating speaker routine and the tradeoffs will be discussed for each algorithm separately. However as an introduction, let's discuss a full featured rotating speaker.

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 rotors. The upper and lower rotors each have a pair of virtual microphones that can be positioned at varying positions (angles) around the rotors. An angle of 0° is loosely defined as the front of the speaker. 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.

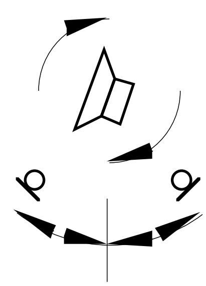


Figure 52 Rotating speaker with virtual microphones

For the rotating speakers, you can control the crossover 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 speed, the effective driver size, and tremolo can be set. The rotation rate 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.

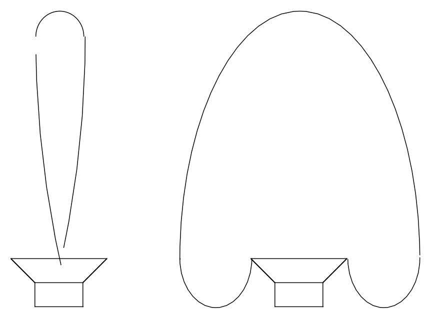


Figure 53 Acoustic beams for (i) low frequency driver and (ii) high frequency driver

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.

VC+Dist+1Rotor 2 models a single rotating speaker in a two-PAU algorithm. In other respects the algorithm is quite full featured and includes the Hammond vibrato/chorus model, distortion, full control of the rotating speaker model (speed, size for Doppler shift, tremolo, acoustic beam width, cabinet resonance) and microphone positions and panning. You get all the features, but only for one driver. The signal does not get split into a high band and low band and passed through separate drivers.

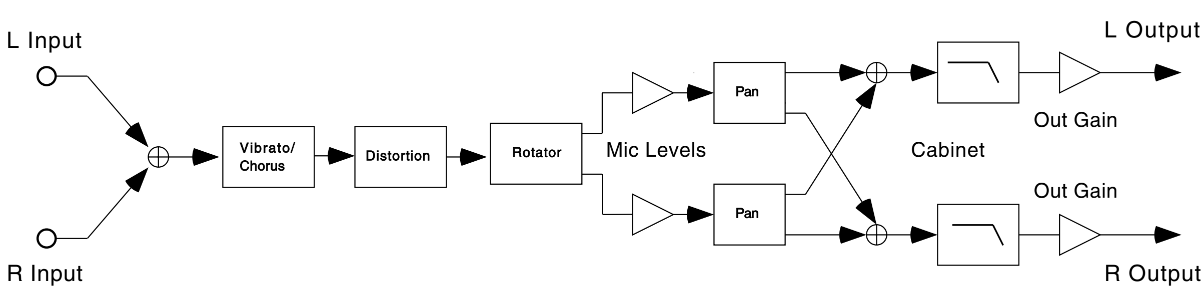


Figure 54 VC+Dist+1Rotor 2

Parameters (VC+Dist+1Rotor 2):

Page 1

		In Gain	Off, -79.0 to 24.0 dB
In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Dist Drive	0.0 to 96.0 dB
Vib/Chor	V1	DistWarmth	8 to 25088 Hz
Roto InOut	In or Out	Cabinet LP	8 to 25088 Hz

Page 2

Gain	Off, -79.0 to 24.0 dB		
Rate	-10.00 to 10.00 Hz		
Size	0 to 250 mm		
Tremolo	0 to 100%		
Beam Width	45.0 to 360.0 deg		

Page 3

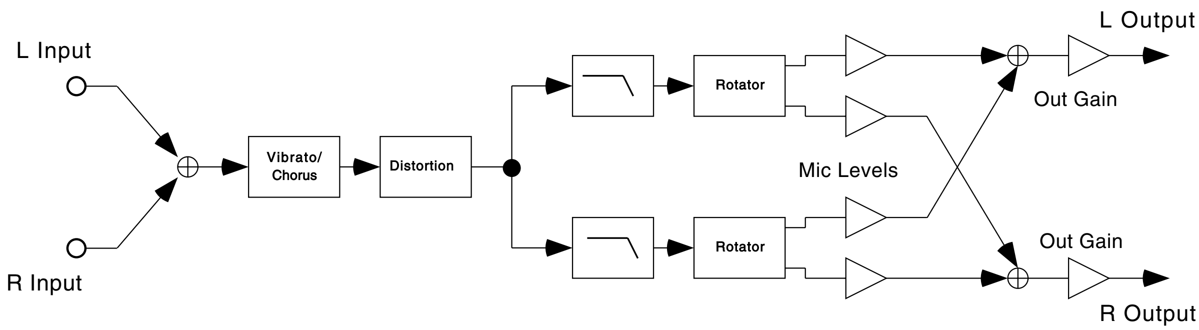
Mic A Pos	-180.0 to 180.0 deg	Mic B Pos	-180.0 to 180.0 deg
Mic A Lvl	0 to 100%	Mic B Lvl	0 to 100%
Mic A Pan	-100 to 100%	Mic B Pan	-100 to 100%

Page 4

Resonate	0 to 100%		
Res Dly	10 to 2550 samp		
Res Xcurs	0 to 510 samp		
Res Phs	0.0 to 360.0 deg		

VC+Dist+HiLoRotr gives you a model of the Hammond vibrato/chorus, distortion and the band splitting for high and low frequency drivers. To pack all this into a two-PAU algorithm, a few sacrifices had to be made to the list of parameters for the rotating speaker model. So what's missing? The resonance controls for the low frequency driver are gone. There is no control of the acoustic beam width for the low driver. The microphone panning is gone and there is a single microphone level control for the A and B microphones. The distortion used is a smaller version of **PolyDistort+EQ**. Even with fewer features, this algorithm gives a convincing Leslie effect while allowing space for more algorithms on other buses.

VC+Dist+HiLoRot2 makes different tradeoffs than **VC+Dist+HiLoRotr**. The distortion is the same as used in **Mono Distortion**. This distortion uses more processor resources than the **PolyDistort+EQ**, so **VC+Dist+HiLoRot2** does not include the acoustic beam width control for either the high or low frequency drivers. The signal flow is the same as for **VC+Dist+HiLoRotr**.



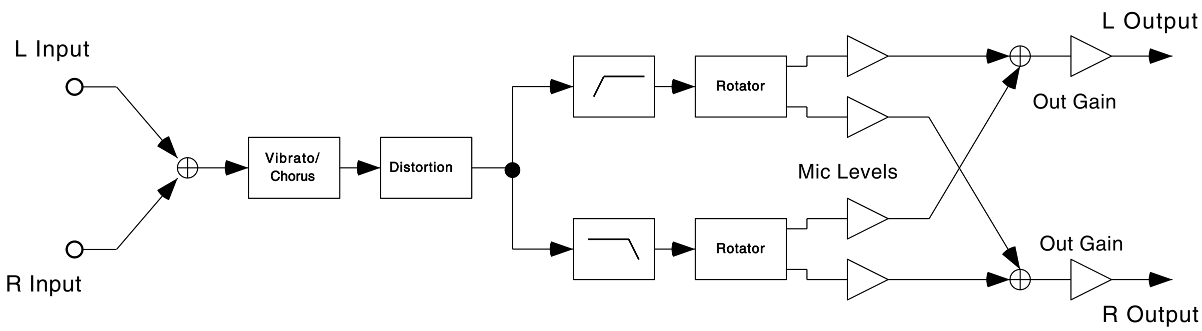


Figure 55 VC+Dist+HiLoRotr and VC+Dist+HiLoRot2

Parameters (VC+Dist+HiLoRotr and VC+Dist+HiLoRot2):

Page 1 (VC+Dist+HiLoRotr)

		In Gain	Off, -79.0 to 24.0 dB
In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Dist Drive	Off, -79.0 to 48.0 dB
Vib/Chor	V1	Dist Curve	0 to 127%
Roto InOut	In or Out	DistLPFreq	8 to 25088 Hz

Page 1 (VC+Dist+HiLoRot2)

In/Out	In or Out	In Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Vib/Chor	V1	Dist Drive	0.0 to 96.0 dB
Roto InOut	In or Out	Dist Warmth	8 to 25088 Hz

Page 2

Xover	8 to 25088 Hz		
Lo Rate	-10 to 10 Hz	Hi Rate	-10 to 10 Hz
Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%
		Hi Beam W	45.0 to 360.0 deg

Page 3

LoMic Lvls	0 to 100%		
LoMicA Pos	-180.0 to 180.0 deg	LoMicB Pos	-180.0 to 180.0 deg
HiMic Lvls	0 to 100%	HiSlow>Fst	0.10 to 10.00 s
HiMicA Pos	-180.0 to 180.0 deg	HiMicB Pos	-180.0 to 180.0 deg

Page 4

		HiResonate	0 to 100%
		Hi Res Dly	10 to 2550 samp
		HiResXcurs	0 to 510 samp
		Res HiPhs	0.0 to 360.0 deg

Rotor 1 is a rotating speaker model on a budget. Its most attractive feature is its small size (one PAU). Obviously a few things had to be scaled back. There is no vibrato/chorus model and no distortion control. There is only a single rotating driver rather than a pair for high and low frequency bands. Aside from these omissions, the rotating speaker model is quite full featured. It includes full control of the rotating speaker including speed, size for Doppler shift, tremolo, acoustic beam width, cabinet lowpass filter and resonance and full microphone control for two microphone positions.

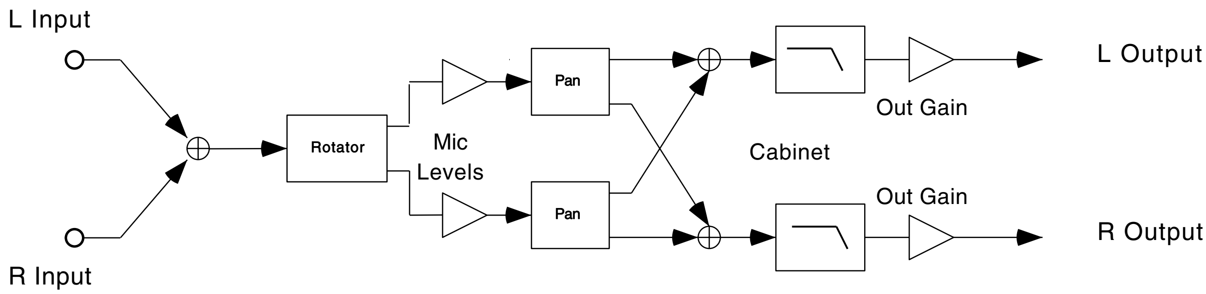


Figure 56 Rotor 1

Parameters (Rotor 1):

Page 1

In/Out	In or Out	In Gain	Off, -79.0 to 24.0 dB
		Out Gain	Off, -79.0 to 24.0 dB
		Cabinet LP	8 to 25088 Hz

Page 2

Gain	Off, -79.0 to 24.0 dB		
Rate	-10 to 10 Hz		
Size	0 to 250 mm		
Trem	0 to 100%		
Beam W	45.0 to 360.0 deg		

Page 3

Mic A Pos	-180.0 to 180.0 deg	Mic B Pos	-180.0 to 180.0 deg
Mic A Lvl	0 to 100%	Mic B Lvl	0 to 100%
Mic A Pan	-100 to 100%	Mic B Pan	-100 to 100%

Page 4

Resonate	0 to 100%		
Res Dly	10 to 2550 samp		
Res Xcurs	0 to 510 samp		
Res Phs	0.0 to 360.0 deg		

VC+Tube+Rotor 4 faithfully models the response and smooth distortion caused by overloading a vacuum tube circuit.

Parameters (VC+Tube+Rotor 4):

Page 1 (VC+Tube+Rotor 4)

In/Out	In or Out	In Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Vib/Chor	V1	Tube Drive	Off, -79.0 to 60.0 dB
Roto InOut	In or Out	Cabinet LP	8 to 25088 Hz

Page 2

Xover	8 to 25088 Hz		
Lo Gain	Off, -79.0 to 24.0 dB	Hi Gain	Off, -79.0 to 24.0 dB
Lo Rate	-10 to 10 Hz	Hi Rate	-10 to 10 Hz
Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%
		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 **Out** the algorithm is bypassed.

In/Out Gain The overall gain or amplitude at the input or 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**, and **V3**, and three choruses **C1**, **C2**, and **C3**.

Roto InOut	When set to **In** the rotary speaker is active; when set to **Out** the rotary speaker is bypassed.
Dist Drive or Tube 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.
Dist Curve	Controls the curvature of the distortion. **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.
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.
DistLPFreq	Controls one-pole lowpass filters in the **PolyDistort+EQ** (in **VC+Dist+HiLoRotr**). Without the lowpass filters, the sound tends to be too bright and raspy. With less distortion drive, less filtering is needed. If you turn off the distortion curve (set to **0**%), you should turn off the lowpass filter by setting it to the highest frequency.
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.
Gain	The gain or amplitude of the signal passing through the rotating speaker.
Rate	The speed of the speaker rotation.
Size	The effective size (radius of rotation) of the rotating speaker in millimeters. Affects the amount of Doppler shift or vibrato of signal.
Tremolo	Controls the depth of tremolo of the signal. Expressed as a percentage of full scale tremolo.
Beam Width	The rotating speaker effect models the acoustic radiation pattern of a speaker ranging 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 speaker is omnidirectional.
Resonate	A simulation of cabinet resonant modes expressed as a percentage. For realism, you should use very low settings.
Res Dly	The number of samples of delay in the resonator circuit in addition to the rotation excursion delay.
Res Xcurs	The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker.
Res Phs	This parameter sets the relative phases the resonators. The angle value in degrees is somewhat arbitrary and you can expect the effect of this parameter to be rather subtle.
Lo Gain	The gain or amplitude of the signal passing through the rotating woofer (low frequency) driver.
Lo Rate	The speed of the woofer rotation.

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 speed of the tweeter rotation.

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 models 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 result in large sample skips (audible as clicks when signal is passing through the effect). There are two pairs of microphones (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 two pairs of microphones (A and B) for high and low frequency drivers.

Mic Pan Left-right panning of the virtual microphone signals. A setting of **-100%** is panned fully left, and **100%** is panned fully right. There are four of these parameters to include two pairs (A and B) for high and low frequency drivers.

LoResonate A simulation of cabinet resonant modes expressed 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 expressed 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.

743 Subtle Distort

Adds small amount of distortion to signal.

PAUs: 1

Use **Subtle Distort** to apply small amounts of distortion to a signal. The distortion characteristic is set with the Curvature and EvenOrders parameters. Increasing Curvature increases the distortion amount while EvenOrders increases the asymmetry of the distortion, adding even distortion harmonics. The distorted signal then is sent through two one-pole lowpass filters and added to the dry input signal. The lowpass filters can reduce any harshness from the raw distortion operation. The Dry In/Out is provided as a utility to audition the distortion signal in the absence of dry signal. Out Gain and Dist Gain can be adjusted together to match the level of the bypassed (dry only) signal. Adding distortion to the dry signal will increase the output level unless Out Gain is reduced.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Dry In/Out	In or Out	Dist Gain	Off, -79.0 to 0.0 dB
Curvature	0 to 100 %	Dist LP A	8 to 25088 Hz
EvenOrders	0 to 100 %	Dist LP B	8 too25088 Hz

In/Out When set to **In** the distortion is active; when set to **Out** the distortion is bypassed.

Dry In/Out Utility parameter to listen to distortion without the dry signal.

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

Dist Gain The gain or amplitude of the distorted signal path prior to passing through the Out Gain adjustment.

Curvature The amount of distortion; none at **0**% and maximum at **100**%.

Even Orders The asymmetry of the distortion (number of even harmonics); none at **0**% and maximum at **100**%.

Dist LP A Frequency of Lowpass Filter A.

Dist LP B Frequency of Lowpass Filter B.

744 Quantize+Alias

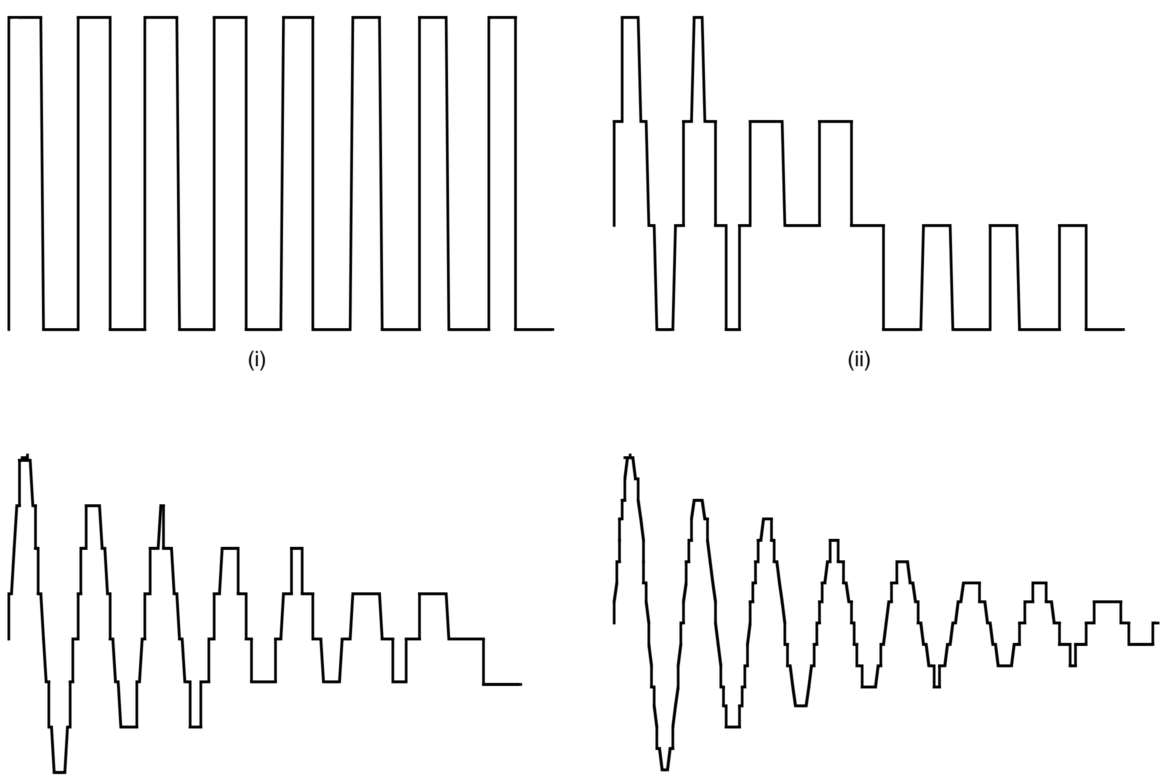
Digital quantization followed by simulated aliasing.

PAUs: 1

The **Quantize+Alias** algorithm offers some of the worst artifacts that digital has to offer! Digital audio engineers will go to great lengths to remove, or at least hide the effects of digital quantization distortion and sampling aliasing. In **Quantize+Alias** we do quite the opposite, making both quantization and aliasing in-your-face effects. The quantizer will give your sound a dirty, grundgy, perhaps industrial sound. The aliasing component simulates the effect of having sampled a sound without adequately band limiting the signal (anti-alias filtering).

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. An 18-bit digital-to-analog converter (DAC) like the one in the K2661 can interpret 262,144 different amplitude levels (2^{18}).

Let's take a look at how finite precision of digital words affects audio signals. The figures below are plots of a decaying sine wave with varying word lengths.



(i) (ii) (iii) (iv)

Figure 57 **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). At **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.

Aliasing is an unwanted artifact (usually!) of digital sampling. It's an established rule in digital sampling that all signal frequency components above half the sampling frequency (the Nyquist rate) must be removed with a lowpass filter (anti-aliasing filter). If frequencies above the Nyquist rate are not removed, you will hear aliasing. A digital sampler cannot represent frequencies above the Nyquist rate, but rather than remove the high frequencies, the sampler folds the high frequencies back down into the lower frequencies where they are added to the original low frequencies. If you were to play a rising pure tone through a sampler without an anti-alias filter, you would hear the tone start to fall when it past the Nyquist rate. The pitch will continue to drop as the input tone's frequency increases until the input tone reaches the sampling rate. The sampled tone would then have reached dc (frequency is 0) and will start to rise again. Usually a lowpass anti-aliasing filter is placed before the sampler to prevent this from happening.

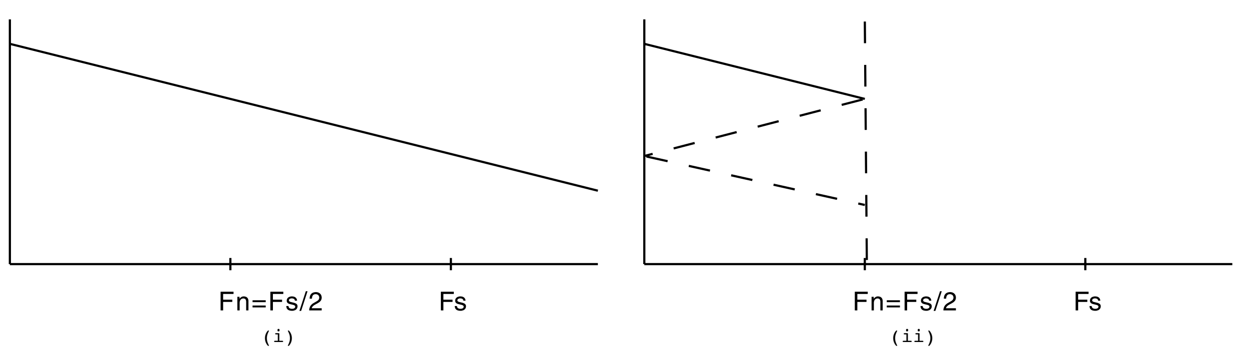


Figure 58 Spectra of (i) an analog signal and the (ii) same signal after sampling without filtering

In the **Quantize+Alias** algorithms, we do not actually sample the incoming signal at a lower rate. Instead we use a special modulation algorithm to simulate the effect of pitches falling when they should be rising. The Pitch (coarse and fine) parameters roughly correspond to setting the Nyquist frequency. Higher pitches result in modulating your input signal with higher frequencies. The LFO Depth parameter changes the strength of the modulation. Larger values of LFO Depth produce a deeper modulation which may be considered analogous to inputting a insufficiently band-limited signal for sampling.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Quant W/D	0 to 100 %wet	DynamRange	0 to 144 dB
Alias W/D	0 to 100 %wet	dc Offset	-79.0 to 0.0 dB
Lowpass	8 to 25088 Hz	Headroom	0 to 144 dB

Page 2

Pitch Crs	8 to 25088 Hz		
Pitch Fine	-100 to 100 ct		
LFO Depth	1 to 49 samp		

In/Out When set to **In**, the quantizer and aliaser are active; when set to **Out**, the quantizer and aliaser are bypassed.

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

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.

Alias W/D Amount of aliaser output signal (wet) relative to aliaser input signal (dry) to send to the final output. The dry signal here is taken to mean the output of the quantizer.

Pitch C
Pitch F Pitch sets the frequency (coarse and fine) at which the input signal is modulated. Higher pitches produce a high frequency modulation.

LFO Depth The depth of the modulation, controlling how strong the modulation sounds. Larger values produce a more extreme modulation effect.

745 Pitcher+MiniVerb

Combination algorithm of Pitcher followed by MiniVerb

PAUs: 2

Pitcher+MiniVerbis **Pitcher** followed by **MiniVerb**. **Pitcher** applies a filter to the signal, the filter having a regular series of peaks in its frequency response which generally imposes a pitch on the input signal. The **MiniVerb** reverb is then applied to the "pitched" signal. See the relevant sections for complete details on these algorithm components.

There are several parameters for controlling the routing and mixing of signals. As might be expected, Wet/Dry sets the level of the wet (**Pitcher+MiniVerb**) signal relative to the main dry input signal. The Mix Pitcher and Mix Reverb parameters set the amounts of the outputs of both of the component effects to send to the main algorithm outputs. The Pch/Dry>Rv parameter sets the amount of pitcher signal to feed to the reverb relative to dry input signal.

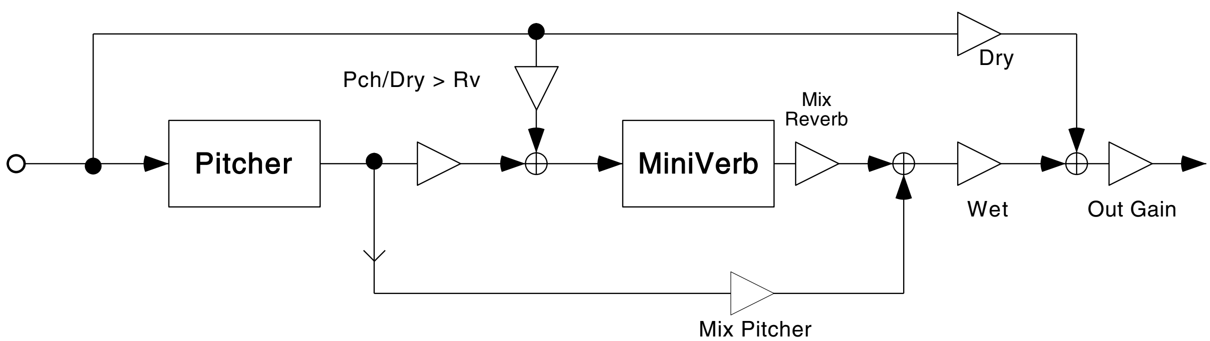


Figure 59 Signal flow of Pitcher+MiniVerb

Parameters:

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Mix Pitcher	0 to 100%		
Mix Reverb	0 to 100%		

Page 2

Pt Pitch	C -1 to G 9		
Pt Offst	-12.0 to 12.0 ST		
Pt Odd Wts	-100 to 100%		
Pt Pair Wts	-100 to 100%		
Pt 1/4 Wts	-100 to 100%		
Pt 1/2 Wts	-100 to 100%		

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Pch/Dry>Rv	0 to 100 %		
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 HFDamp	8 to 25088 Hz
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms

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.

Mix Pitcher Adjusts the amount of the pitcher effect that is mixed together as the algorithm wet signal. Negative values polarity invert that particular signal.

Mix Reverb Adjusts the amount of the reverb effect that is mixed together as the algorithm wet signal. Negative values polarity invert that particular signal.

Pt Pitch The fundamental pitch imposed upon the input. Values are in MIDI note numbers.

Pt Offst An offset from the pitch frequency in semitones. This is also available for adding an additional continuous controller mod like pitch bend.

Pt Odd Wts
Pt Pair Wts
Pt 1/4 Wts
Pt 1/2 Wts
These parameters control the exact shape of the frequency response of **Poly 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 in the section on **Pitcher**.

Pch/Dry->Rv This parameter controls how much of the pitcher effect is mixed with dry and fed into the reverb effect. This control functions like a wet/dry mix, where **0%** is completely dry and **100%** is pitcher effect only.

Rv Time The reverb time displayed is accurate for normal settings of the other parameters (Rv HF Damp = **25088 kHz**, and Rv DiffScl, Rv SizeScl and Rv Density = **1.00x**). Changing Rv Time to **Inf** creates an infinitely sustaining reverb.

Rv 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 Rv Types will sound best with shorter Rv Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you may not modulate it.)

Rv DiffScl A multiplier that affects the diffusion of the reverb. At **1.00x**, the diffusion will be the normal, carefully adjusted amount for the current Rv Type. Altering this parameter will change the diffusion from the preset amount.

Rv SizeScl A multiplier that changes the size of the current room. At **1.00x**, the room will be the normal, carefully tweaked size of the current Rv 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).

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

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

Rv PreDlyL/R The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer predelays 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.

746 Reverb+Compress

A reverb and compressor in series.

PAUs: 2

Reverb+Compress is configured as a reverb followed by a compressor. The reverbs used are the same as **MiniVerb**. The compressor is a soft-knee compressor and can be configured as a feed-forward or feedback compressor.

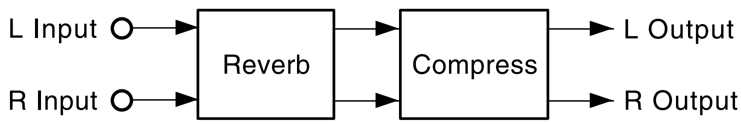

Figure 60 Simplified block diagrams of Reverb+Compress

The main control for the reverbs is the Rv Type parameter. Rv Type changes the structure of the algorithms to simulate many carefully crafted room types and sizes. Spaces characterized as booths, small rooms, chambers, halls and large spaces can be selected. For a complete discussion on the reverbs see the sections on **MiniVerb**.

The compressor reduces 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.

In the soft-knee compressor there is a gradual transition from compressed to unity gain.

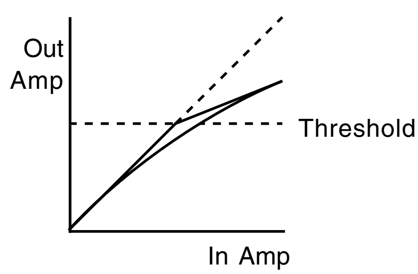

Figure 61 Soft-Knee compression characteristics

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 overshoot 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 behavior, 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: CompSmooth. 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 displays the amount of gain reduction applied to the signal as a result of compression.

Parameters:

Page 1

In/Out	In or Out	ReverbGain	Off, -79.0 to 24.0 dB
Reverb W/D	0 to 100 %wet	Rv Time	0.5 to 30.0 s, Inf
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms
Rv HFDamp	8 to 25088 Hz	CompIn/Out	In or Out

Page 2

Rv Type	Hall1, etc.	Rv DiffScl	0.00 to 2.00 x
		Rv SizeScl	0.00 to 4.00 x
		Rv Density	0.00 to 4.00 x

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Comp Atk	0.0 to 228.0 ms	Comp Ratio	1.0:1 to 100.0:1, Inf:1
Comp Rel	0 to 3000 ms	Comp Thres	-79.0 to 0.0 dB
CompSmooth	0.0 to 228.0 ms	CompMakeUp	Off, -79.0 to 24.0 dB
CompSigDly	0.0 to 25.0 ms	FdbkComprs	In or Out
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

In/Out	When set to **In** the overall algorithm is active; when set to **Out** the algorithm is bypassed.
ReverbW/D	This is a simple mix of the reverb input (dry) with the reverb output (wet) to produce the final reverb output.
ReverbGain	An overall level control of the reverb's output (applied after the reverb Wet/Dry mix).
Rv HFDamp	Reduces high frequency components of the reverb above the displayed cutoff frequency. Removing higher reverb frequencies can often make rooms sound more natural.
Rv PreDlyL/R	The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer predelays 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.
CompIn/Out	When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.
Rv 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 Rv Types will sound best with shorter Rv Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you may not modulate it.)
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.
CompSmooth	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.
CompSigDly	The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain predelay). This allows the compression to appear to take effect just before the signal actually rises.
Comp Ratio	The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.
Comp Thres	The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.
CompMakeUp	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.
FdbkComprs	A switch to set whether the compressor side chain is configured for feed-forward (**Out**) or feedback (**In**).

781 St Chorus+Delay
784 St Flange+Delay

Combination effect algorithms using time/frequency units instead of tempo

PAUs: 1 or 2

The algorithms listed here are identical in most respects to combination effects elsewhere documented. For example, **St Chorus+Delay** is closely based on **Chorus+Delay**. The difference for algorithms with "St" in the name is that they use stereo controls (ganged controls) rather than dual mono controls for the chorus and flange components of the algorithms.

790 Gate+Cmp[EQ]+Vrb

Combination algorithm designed for vocal processing.

PAUs: 4 each

This algorithm is provided with vocal processing in mind. It includes a gate followed by a compressor and a reverb. Equalization is included as part of the compressor's side-chain processing. Side-chain equalization allows some interesting processing possibilities including "de-essing" (by boosting the treble in the side-chain). For each configuration of compressor and EQ, the EQ includes bass, treble and mid controls (gain and frequency for each plus width for the mid EQ).

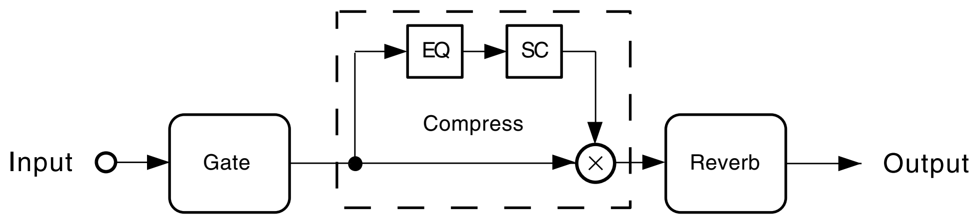

Figure 62 Gate+Cmp[EQ]+Vrb

The gate (same gate as **Gate**) allows you to cut out noise during vocal silence. You must decide whether to gate based on left or right channels or to gate based on both channels (average magnitude). Both the gate and compressor have their own side-chain processing paths. For both the gate and compressor, side-chain input may be taken from either the left or right channels, or the average signal magnitude of the left and right channels may be selected using the GateSCInp or CompSCInp parameters.

The reverb is the same as used in **MiniVerb**. You will find all the same controls and room settings. In the FXPreset editor, you will have to scroll with the **more>** soft button to find the **PARAM5** soft button containing the reverb parameters.

Parameters:

Page 1

GateIn/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
GateSCInp	L, R, (L+R)/2	CompIn/Out	In or Out
		CompSCInp	L, R, (L+R)/2
		FdbkComprs	In or Out

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Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	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

Page 3

Comp Atk	0.0 to 228.0 ms	Comp Ratio	1.0:1 to 100:1, Inf:1
Comp Rel	0 to 3000 ms	Comp Thres	-79.0 to 0.0dB
CompSmooth	0.0 to 228.0 ms	CompMakeUp	Off, -79.0 to 24.0 dB
CompSigDly	0.0 to 25.0ms		

Page 4

CmpSCBassG	-79.0 to 24.0 dB	CmpSCTrebG	-79.0 to 24.0 dB
CmpSCBassF	8 to 25088 Hz	CmpSCTrebF	8 to 25088 Hz
CmpSCMidG	-79.0 to 24.0 dB	Comp SC EQ	In or Out
CmpSCMidF	8 to 25088 Hz		
CmpSCMidW	0.010 to 5.000 oct		

Page 5

Reverb W/D	0 to 100 %wet		
Rv Type	Hall1, etc.		
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	8 to 25088 Hz
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms

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

GateIn/Out When set to **In** the gate is active; when set to **Out** the gate is bypassed.

GateSCInp Select the input source channel for gate side-chain processing—left, right or both. For both (L+R)/2 the averaged magnitude is used.

CompIn/Out When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.

CompSCInp Select the input source channel for compressor side-chain processing**—Left**, **Right** or **Both**. For both (L+R)/2 the averaged magnitude is used.

FdbkComprs A switch to set whether the compressor side-chain is configured for feed-forward (**Out**) or feedback (**In**).

Gate Thres The 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 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 time for the gate to ramp from open to closed (reverse if Gate Duck is **On**) after the gate timer has elapsed.

GateSigDly 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.

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.

CompSmooth A lowpass filter in the compressor side-chain signal path. It is intended to smooth the output of the compressor's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

CompSigDly The time in ms by which the input signal should be delayed with respect to compressor side-chain processing (i.e. side-chain predelay). This allows the compression to appear to take effect just before the signal actually rises.

Comp Ratio The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

Comp Thres The compressor threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

CompMakeUp
A gain or amplitude control provided to offset gain reduction due to compression.

The EQ parameters with names starting with **CmpSC** refer to EQ filters in the side-chain processing path of **Gate+Cmp[EQ]+Vrb**.

CmpSCBassG, 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.

CmpSCBassF, Bass Freq The center frequency of the bass shelving filter in intervals of one semitone.

CmpSCTrebG, 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.

CmpSCTrebF, Treb Freq The center frequency of the treble shelving filters in intervals of one semitone.

CmpSCMidG, Mid Gain The amount of boost or cut that the 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.
CmpSCMidF, Mid Freq	The center frequency of the parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.
CmpSCMidW, Mid Width	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.
Reverb W/D	A simple mix of the reverb sound with the dry (compressed) sound.
Rv PreDlyL/R	The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer predelays 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.
Rv Time	The reverb time displayed is accurate for normal settings of the other parameters (HF Damping = **25088 kHz**, and Rv DiffScl, Rv SizeScl and Rv Density = **1.00x**). Changing Rv Time to **Inf** creates an infinitely sustaining reverb.
Rv 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 Rv Types will sound best with shorter Rv Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you may not modulate it.)
Rv HF Damp	Reduces high frequency components of the reverb above the displayed cutoff frequency. Removing higher reverb frequencies can often make rooms sound more natural.
Rv DiffScl	A multiplier which affects the diffusion of the reverb. At **1.00x**, the diffusion will be the normal, carefully adjusted amount for the current Rv Type. Altering this parameter will change the diffusion from the preset amount.
Rv SizeScl	A multiplier which changes the reverb size of the current room. At **1.00x**, the room will be the normal, carefully tweaked size of the current Rv 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).
Rv 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 Rv Type. Altering this parameter will change the density of the reverb, which may color the room slightly.

792 Gate+TubeAmp

Combination algorithm designed for guitar processing.

PAUs: 3

This algorithm is provided with guitar processing in mind. It sends the signal through a gate, tone controls, tube distortion and cabinet simulation or EQ section. Also depending on the algorithm selected, the signal may pass through one or more of compressor, equalization, chorus, flange, moving delay or reverb. The algorithm is mono, though the chorus or flange can provide stereo spreading at the output.

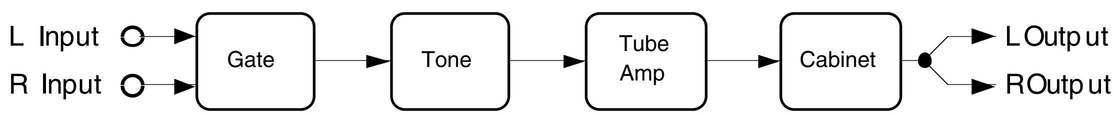


Figure 63 Gate+TubeAmp

The gate (same gate as **Gate**) allows you to cut out noise during silence. Both the gate and compressor have their own side-chain processing paths, and a number of signal routing options for side-chain processing are provided. The gate side-chain input may be taken from either the left or right channels, or the average signal magnitude of the left and right channels may be selected with the GateSCInp parameter. Also you may choose to gate the sum of left and right channels or just one of the channels with the Gate Chan parameter. Since the effect is mono, if you gate only one channel (left or right), then that channel will be sent to the next stage of the effect, and the channel that is not selected will be discarded. If you choose both (L+R)/2, the sum (mix) of both channels will be used for further processing.

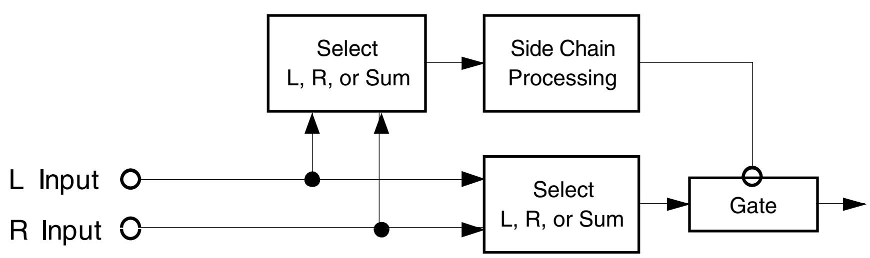


Figure 64 Gate routings

Each of the guitar combi algorithms contains a tone control and tube distortion model. The 3-band tone control authentically recreates the response in many guitar preamps based on real measurements collected by Kurzweil engineers. It is adjusted with the Bass Tone, Mid Tone, and Treb Tone controls with values ranging from **0** to **10** commonly found on many guitar amps. The flattest frequency response is obtained by setting Mid Tone to **10**, and both Bass and Treb Tone controls to **0**. The Tube Drive parameter faithfully model the response and smooth distortion caused by overloading a vacuum tube circuit.

Following the tube distortion is cabinet simulation or an EQ section with parametric bass, treble and mid-range equalization filters. The cabinet simulator models the responses of various types of mic'd guitar cabinets. The preset can be selected using the Cab Preset parameter. The presets are described below.

Basic	Flat response from 100 Hz to 4 khz with 24dB/oct rolloffs on each end
Lead 12	Open back hard American type with one 12” driver
2x12	Closed back classic American type with two 12” drivers
Open 12	Open back classic American type with one 12” driver
Open 10	Open back classic American type with one 10” driver
4x12	Closed back British type with four 12” drivers
Hot 2x12	Closed back hot rod type with two 12” drivers
Hot 12	Open back hot rod type with one 12” driver

Parameters (Gate+TubeAmp):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
GateIn/Out	In or Out		
GateSCInp	L, R, (L+R)/2		
Gate Chan	L, R, (L+R)/2		

Page 2

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	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

Page 3

Bass Tone	0.0 to 10.0	Tube Drive	Off, -79.0 to 60.0 dB
Mid Tone	0.0 to 10.0	Warmth	8 to 25088 Hz
Treb Tone	0.0 to 10.0	Cab Preset	Open 12, ...

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 entire algorithm.

GateIn/Out When set to **In** the gate is active; when set to **Out** the gate is bypassed.

GateSCInp Select the input source channel for gate side-chain processing—left, right or both. For both (L+R)/2 the averaged magnitude is used.

Gate Chan Select which input channel will receive gate processing—left, right or mix. This selects the mono input for the algorithm.

Gate Thres The signal level in dB required to open the gate (or close the gate if Ducking is **On**).

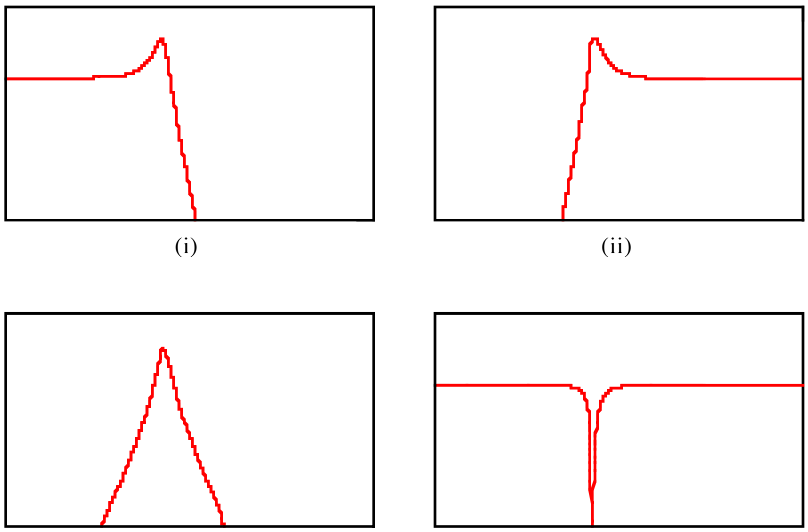
Gate Duck	When set to **Off**, the gate opens when the signal rises above threshold and closes when the gate time expires. When set to **On**, the gate closes when the signal rises above threshold and opens when the gate time expires.
Gate Time	The time in seconds that the gate will stay fully on after the signal envelope rises above threshold. The gate timer is started or restarted whenever the signal envelope rises above threshold.
Gate Atk	The 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 time for the gate to ramp from open to closed (reverse if Gate Duck is **On**) after the gate timer has elapsed.
GateSigDly	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.
Bass Tone **Mid Tone** **Treb Tone**	Adjusts the three bands of tone control integrated with the distortion drive circuit. Flattest response is obtained by setting Mid Tone to **10.0** and both Bass Tone and Treb Tone to 0.0.
Tube Drive	Adjusts the gain into the distortion circuit. Higher values produce more distortion.
Cab Preset	Eight preset cabinets have been created based on measurements of real guitar amplifier cabinets. The presets are **Basic**, **Lead 12**, **2x12**, **Open 12**, **Open 10**, **4x12**, **Hot 2x12**, and **Hot 12**.
Warmth	Adjusts a 1 pole (6dB/oct) lowpass filter applied after distortion.

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).



(i) (ii) (iii) (iv)

Figure 10-65 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.

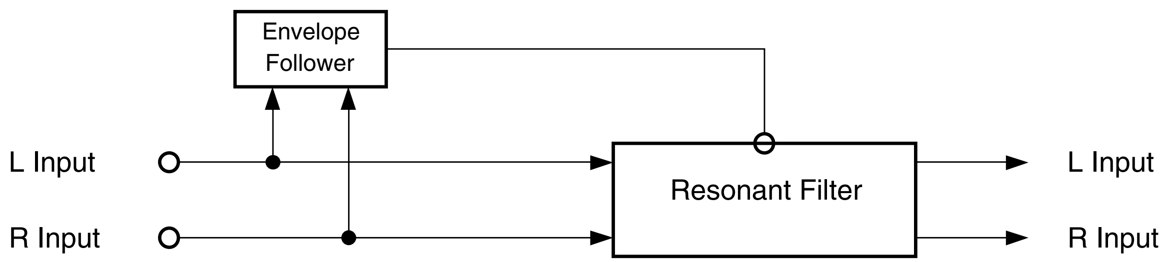


Figure 10-66 Block diagram of envelope following filter

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).

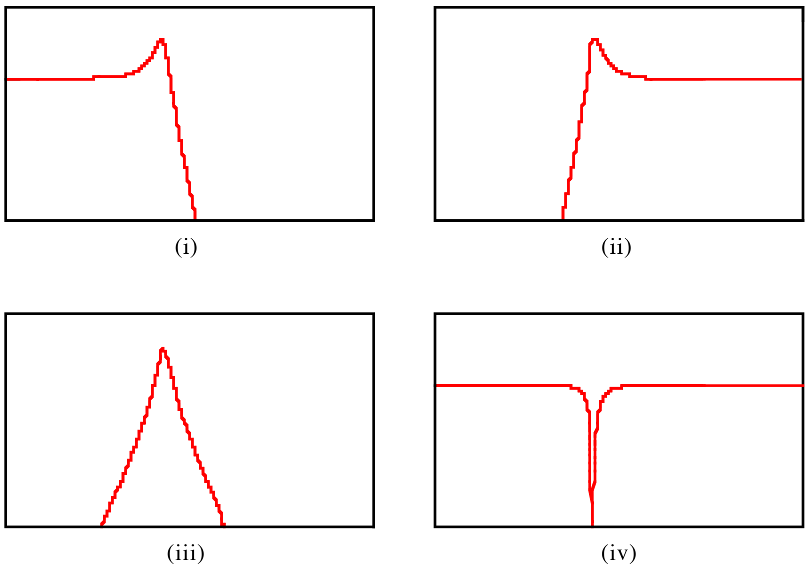


Figure 10-67 Resonant Filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

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.

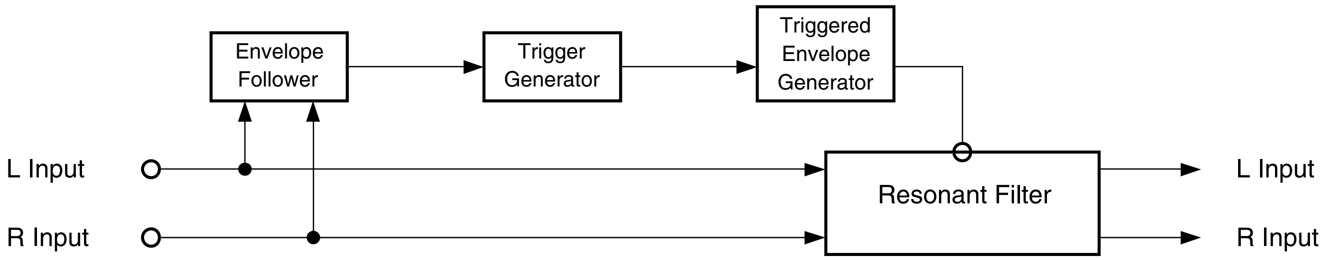


Figure 10-68 Block diagram of Triggered Envelope Filter

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).

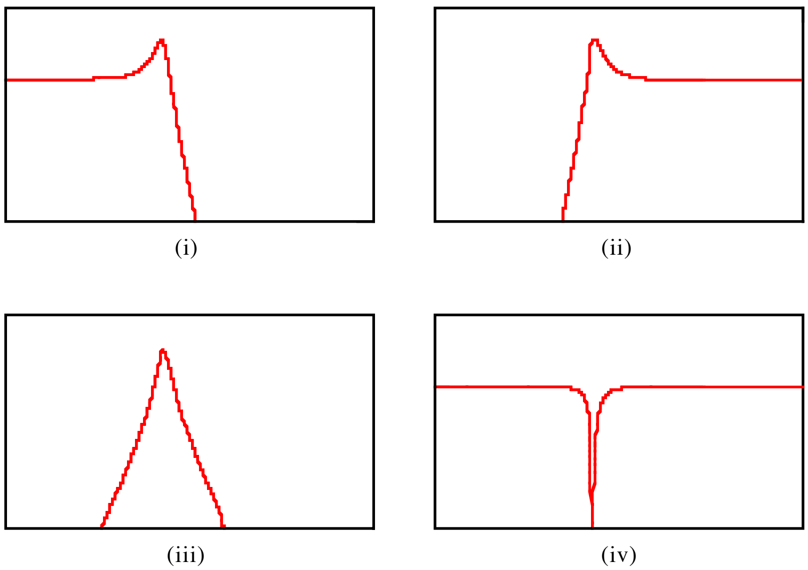


Figure 10-69 Resonant Filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

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.

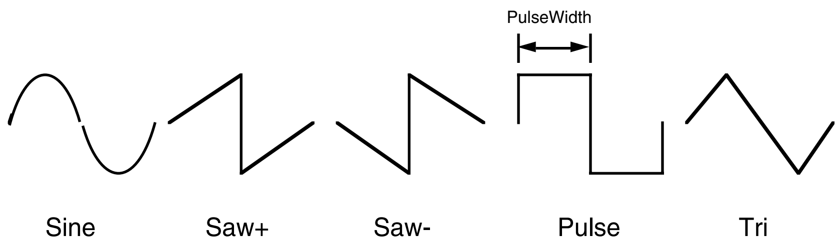


Figure 10-70 Configurable Wave Shapes

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).

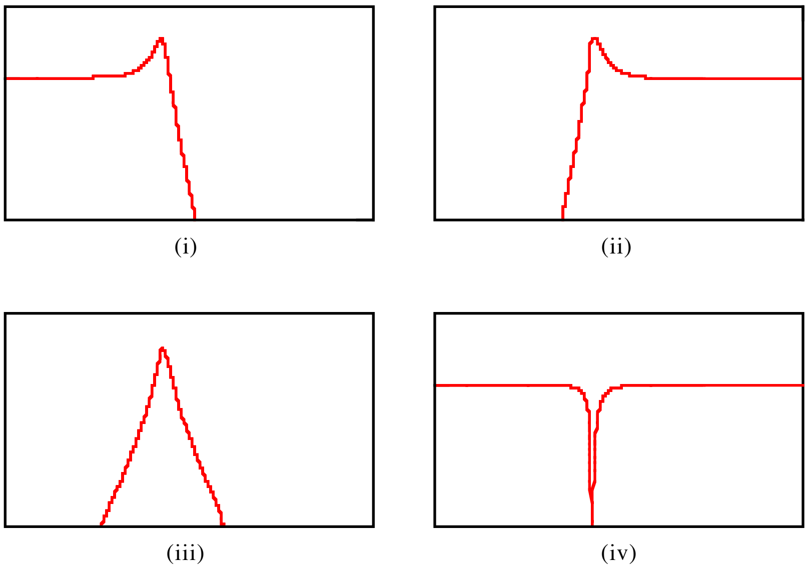


Figure 10-71 Resonant Filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

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.

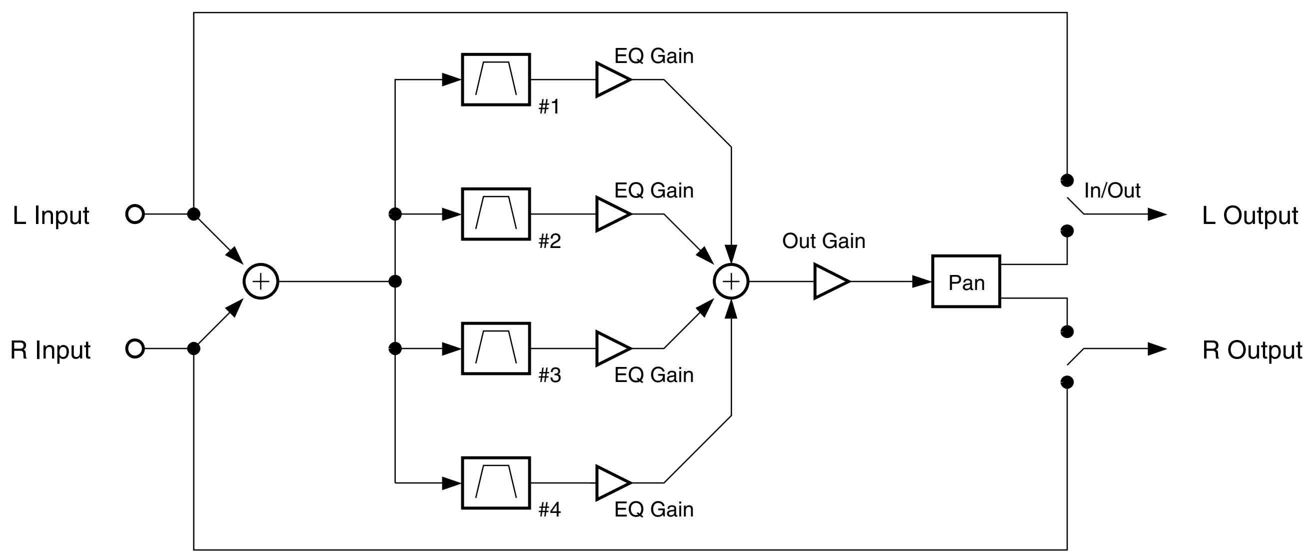


Figure 10-72 Mono EQ Morpher (EQ Morpher is similar)

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.

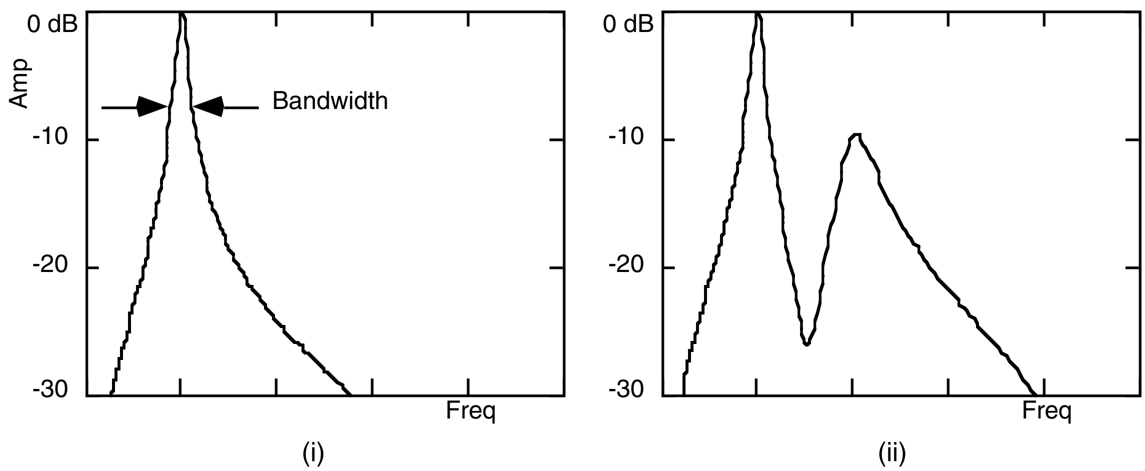


Figure 10-73 Frequency response of (i) a single bandpass filter; (ii) the sum of two bandpass filters

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):

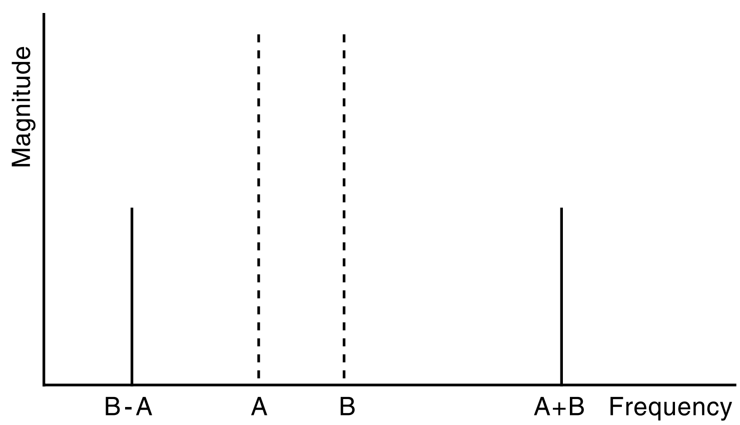


Figure 10-74 Result of Modulating Two Sine Waves A and B

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:

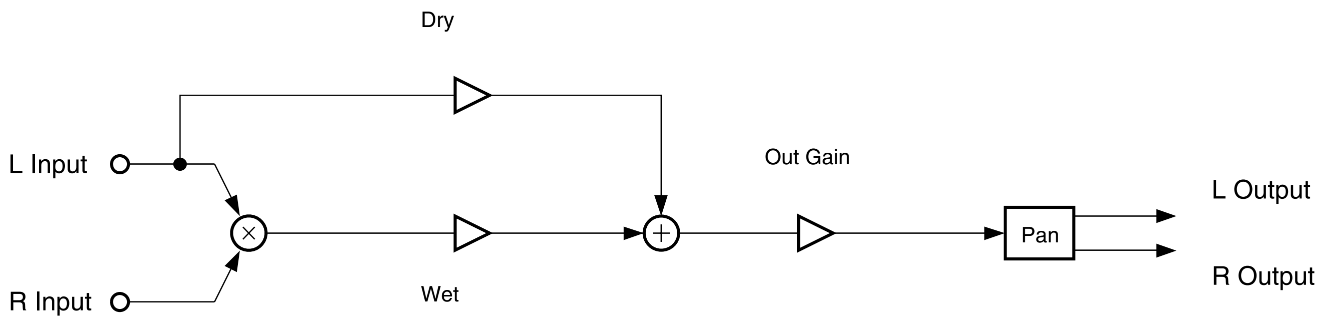


Figure 10-75 "L*R" Mode Ring Modulator

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 10-76).

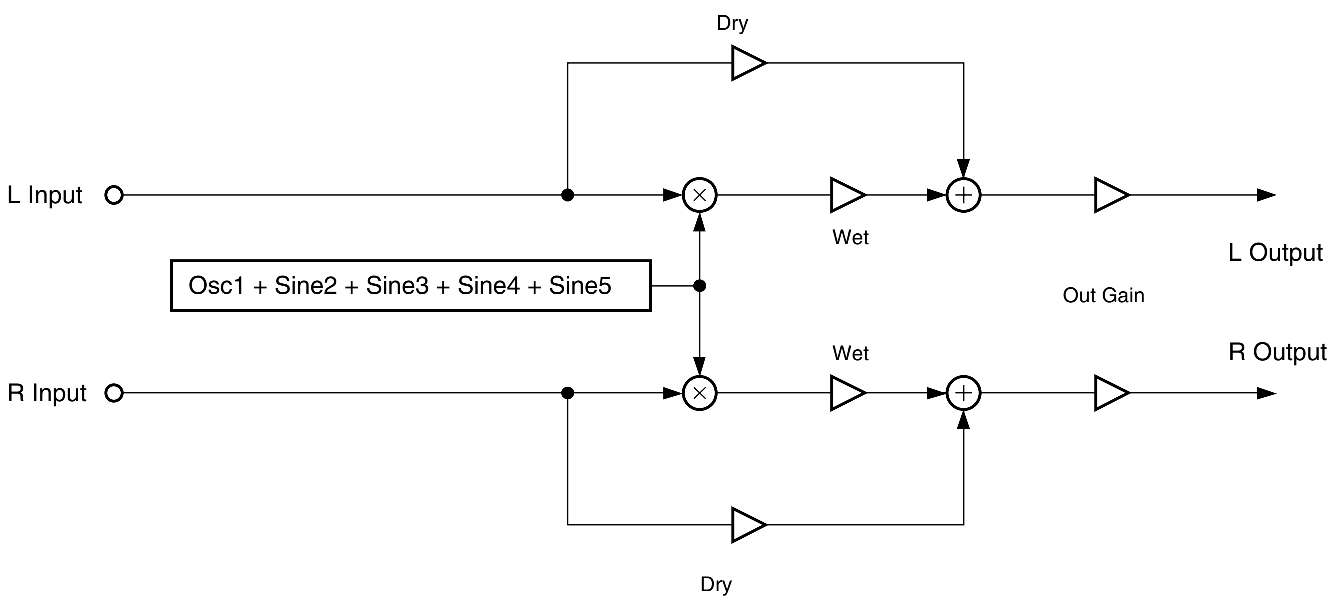


Figure 10-76 "Osc" Mode Ring Modulator

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.

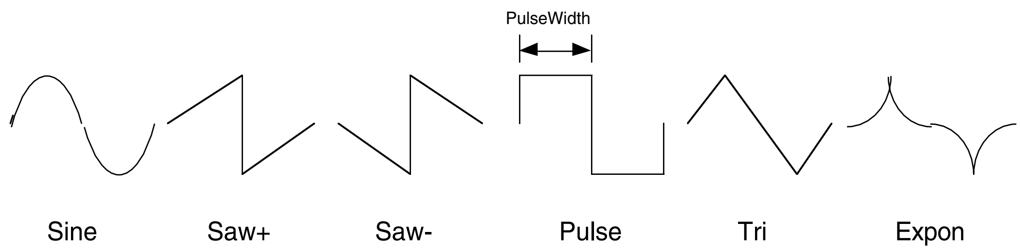


Figure 10-77 Configurable Wave Shapes

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].

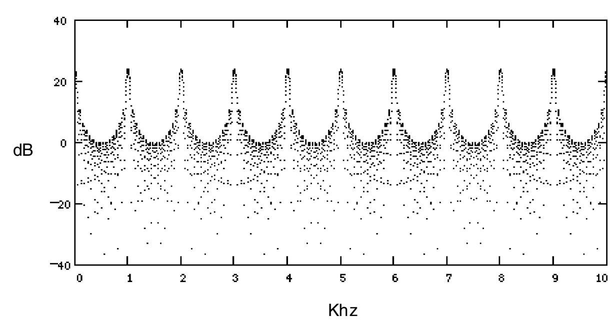


Figure 10-78 [100, 100, 100, 100]

In Figure 10-78, all peaks are exact multiples of the fundamental frequency set by the Pitch parameter. This setting gives the most "pitchiness" to the output.

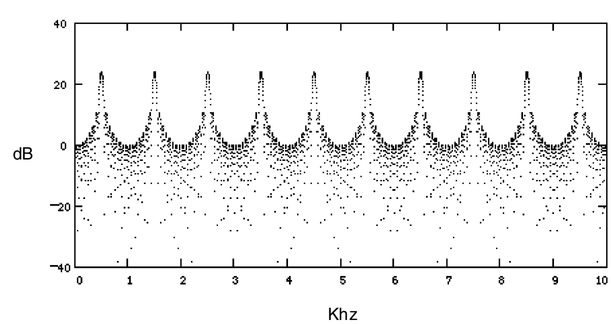


Figure 10-79 [-100, 100, 100, 100]

In Figure 10-79, peaks are odd multiples of a frequency one octave down from the Pitch setting. This gives a hollow, square-wavey sound to the output.

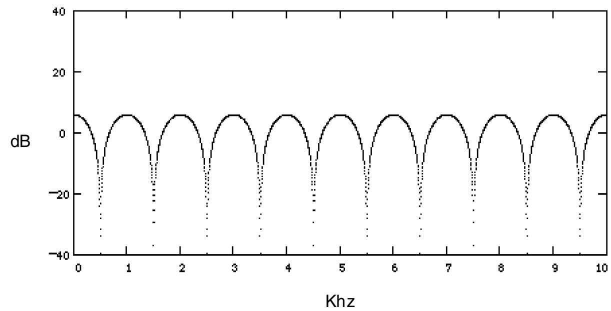


Figure 10-80 [100, 0, 0, 0]

In Figure 10-80, there are deeper notches between wider peaks

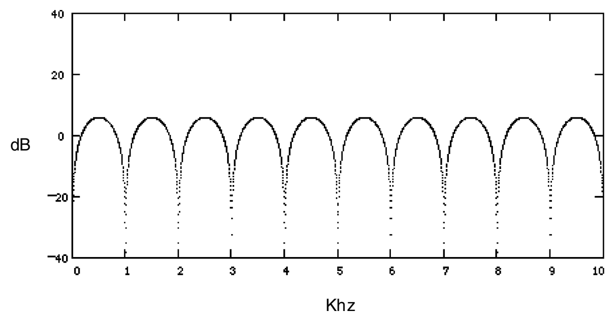


Figure 10-81 [-100, 0, 0, 0]

In Figure 10-81, there are peaks on odd harmonic multiples and notches on even harmonic multiples of a frequency one octave down from the Pitch setting.

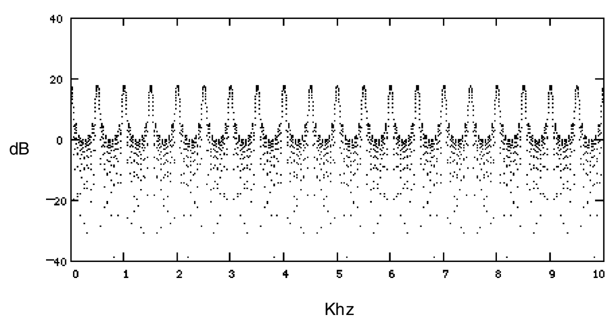


Figure 10-82 [0, 100, 100, 100]

Figure 10-82 is like [100,100,100,100], except that all the peaks are at (all) multiples of half the Pitch frequency.

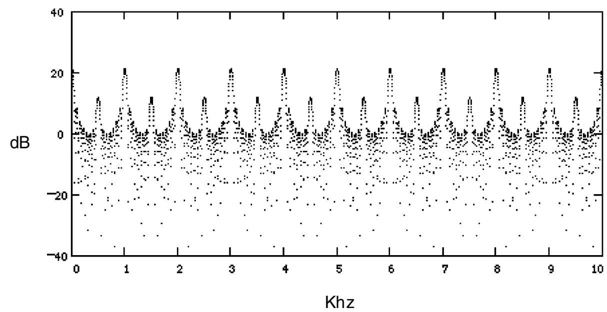


Figure 10-83 [50,100,100,100]

Figure 10-83 is halfway between [0,100,100,100] and [100,100,100,100].

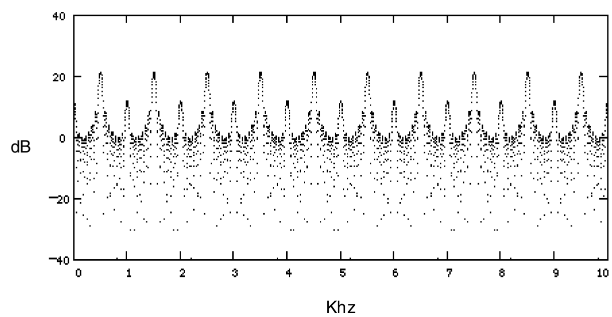


Figure 10-84 [-50,100,100,100]

Figure 10-84 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.

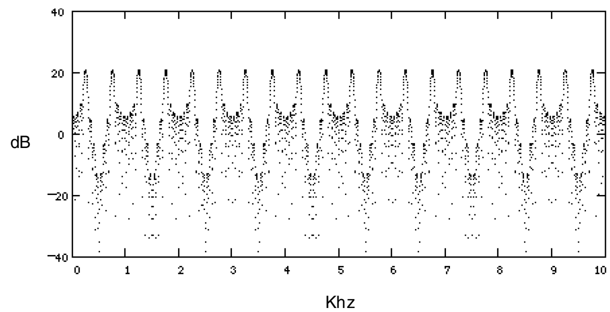


Figure 10-85 [100, -100, 100, 100]

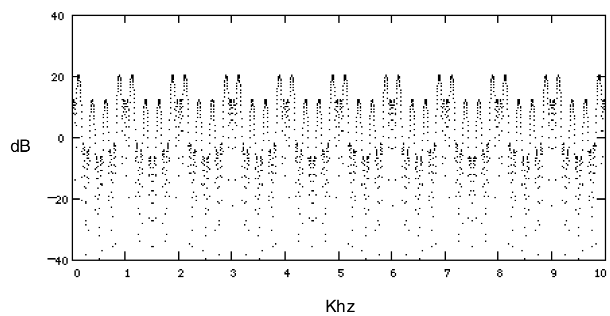


Figure 10-86 [100, 100, -100, 100]

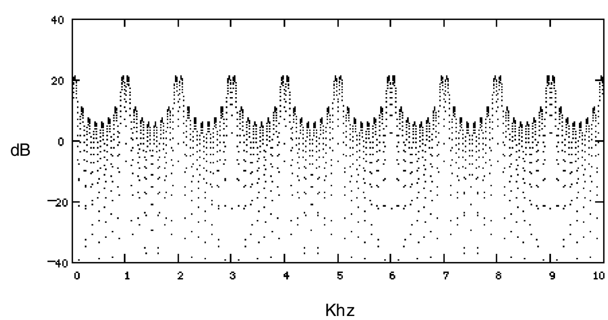


Figure 10-87 [100, 100, 100, -100]

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.

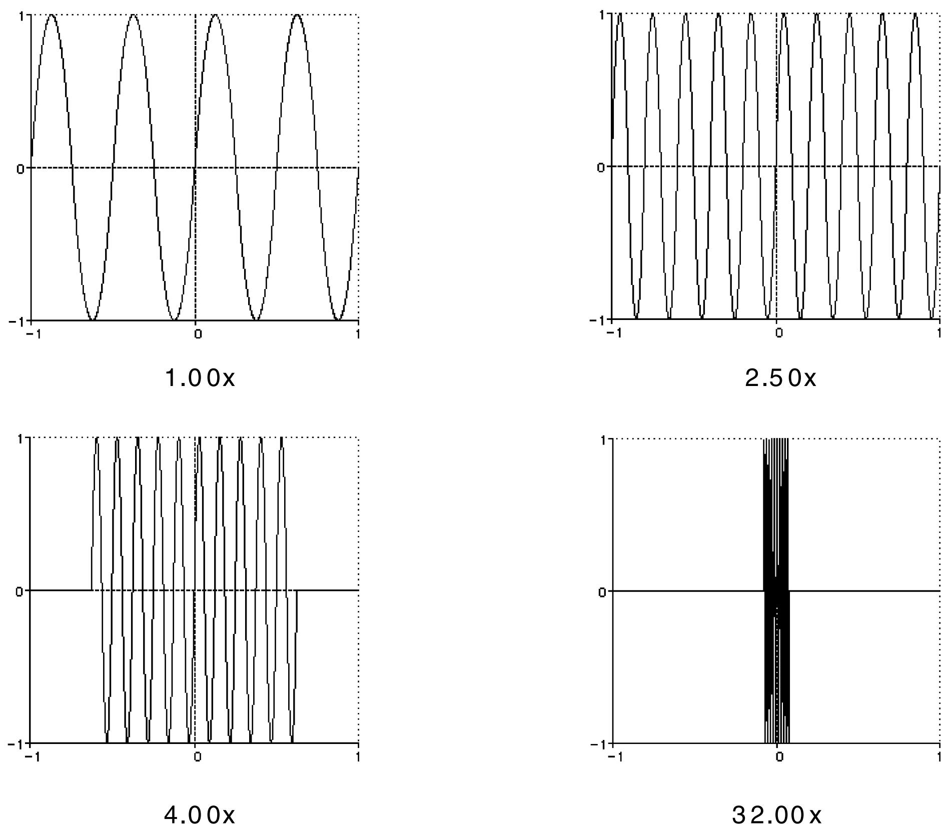


Figure 10-88 Super Shaper: Four Values of the Amount Parameter

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.)

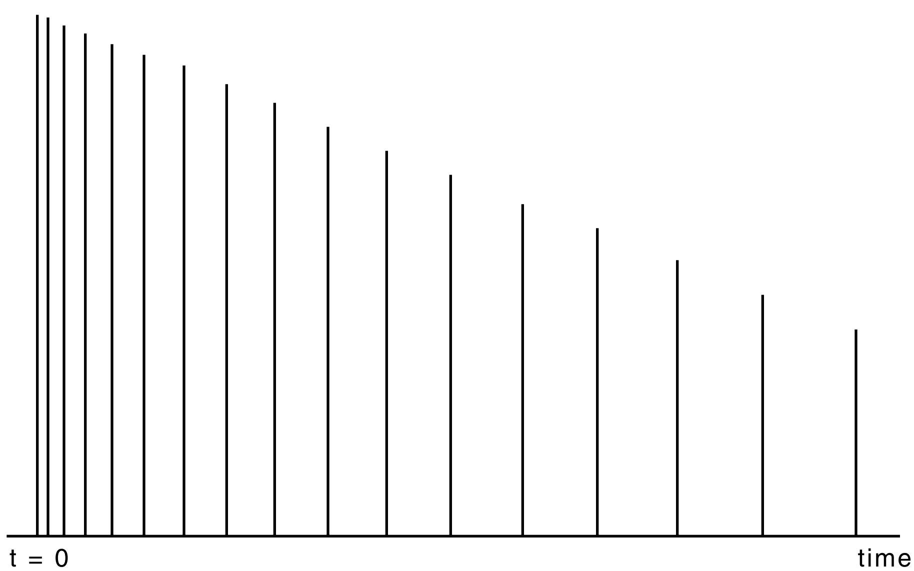


Figure 10-89 Simplified Impulse Response of LaserVerb

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 Contour 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.

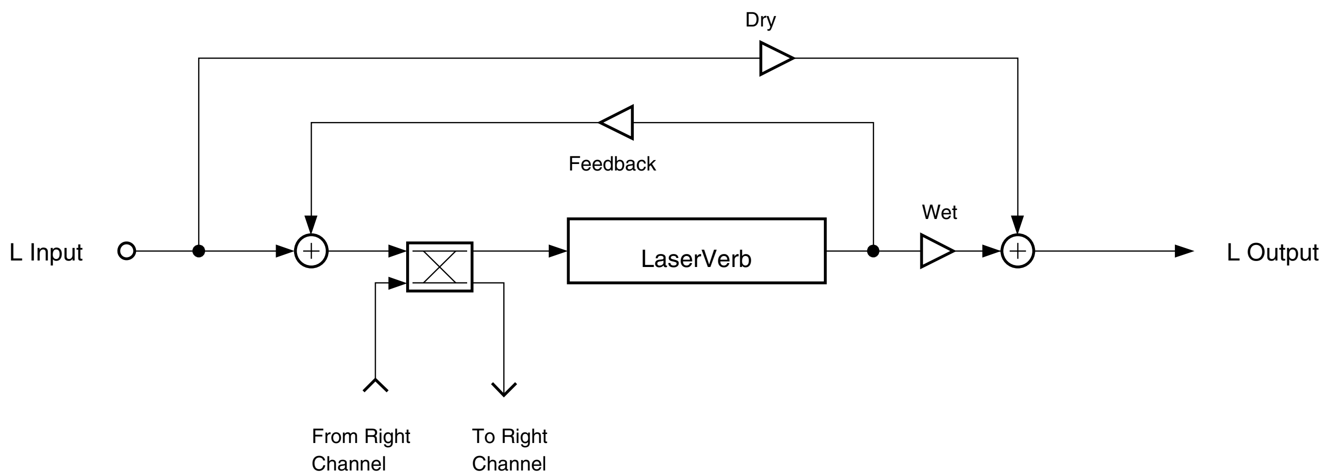


Figure 10-90 LaserVerb

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.

914 Revrse LaserVerb

A bizarre reverb which runs backwards in time.

PAUs: 4

Revrse LaserVerb is a mono effect that simulates the effect of running the **LaserVerb** in reverse. When you play a sound through the algorithm, it starts out relatively diffuse then builds to the final "hit." Since KDFX cannot break the universal rules of causality (sorry, KDFX doesn't know what you are about to play!), there can be a significant delay between what you play and when you hear it. In addition to the normal Wet/Dry control, with the Rvrs W/D, the dry signal is considered to be the delayed "hit" signal.

Revrse LaserVerb is **LaserVerb** in reverse, so when it is fed an impulsive sound such as a snare drum, it plays the impulse back as a delayed train of closely spaced impulses, and as time passes, the spacing between the impulses gets closer until they coalesce at the "hit." The close spacing of the impulses produces a discernible buzzy pitch which gets higher as the impulse spacing decreases. The following figure is a simplified representation of the **Revrse 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.)

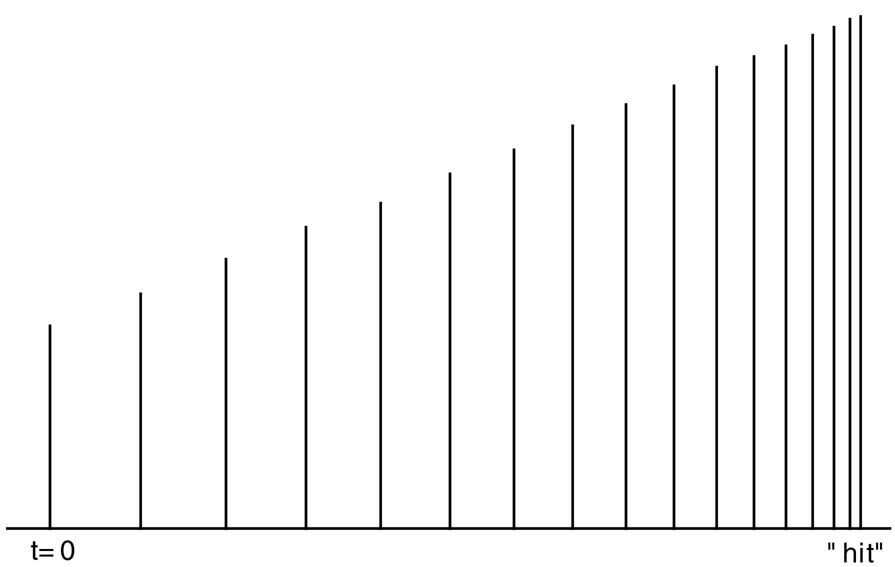


Figure 91 Simplified impulse response of Revrse LaserVerb

With appropriate parameter settings this effect produces an ascending buzz or whine. The ascending buzz is most prominent when given an impulsive input such as a drum hit. To get the ascending buzz, start with about half a second of delay and set the Contour parameter to a high value (near **100%**). The Contour parameter controls the overall shape of the **LaserVerb** impulse response. At high values the response builds up slowly to the "hit." As the Contour value is reduced, the response starts out lower and rises more rapidly to the "hit."

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.

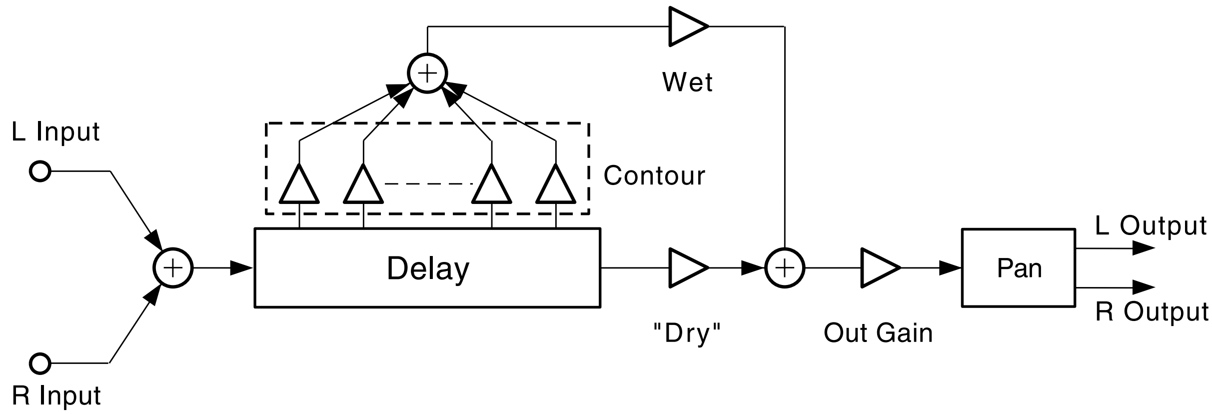


Figure 92 Revrse LaserVerb

Parameters:

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Rvrs W/D	0 to 100 %wet	Pan	-100 to 100 %

Page 2

Dly Coarse	0 to 5000 ms	Contour	0.0 to 100.0 %
Dly Fine	-20.0 to 20.0 ms		
Spacing	0 to 200 samp		

Wet/Dry The amount of reverbed (wet) signal relative to unaffected (dry) signal.

Rvrs W/D A special wet/dry control in which the “dry” signal is in fact delayed so that it is the last sound to be sent to the output, as if the LaserVerb is being played in reverse.

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

Pan The left and right inputs get summed to mono, the mono signal passes through the **Revrse 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 **5** seconds. Lengthening the delay will increase the duration or decay time of the reverb.

Dly Fine The delay fine adjust is added to the delay coarse adjust to provide a delay resolution down to **0.2** ms.

Spacing Determines the starting pitch of the ascending buzz and how fast it ascends. The Spacing parameter sets the initial separation of impulses in the impulse response and subsequent rate of decreasing 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 builds to a higher frequency than for higher Spacing settings.

Contour	Controls the overall envelope shape of the reverb. When set to a high value, sounds start at a high level and build slowly to the final "hit." As the control value is reduced, sounds start lower and build rapidly to the final "hit."

915 Gated LaserVerb

The LaserVerb algorithm with a gate on the output.

PAUs: 3

Gated LaserVerb is **LaserVerb Lite** with a gate on the output. For a detailed explanation of **LaserVerb** see the section for **LaserVerb Lite**. The gate controls are covered under **Gate**. Signal routings between the inputs, the **LaserVerb**, the gate, and the outputs are described here.

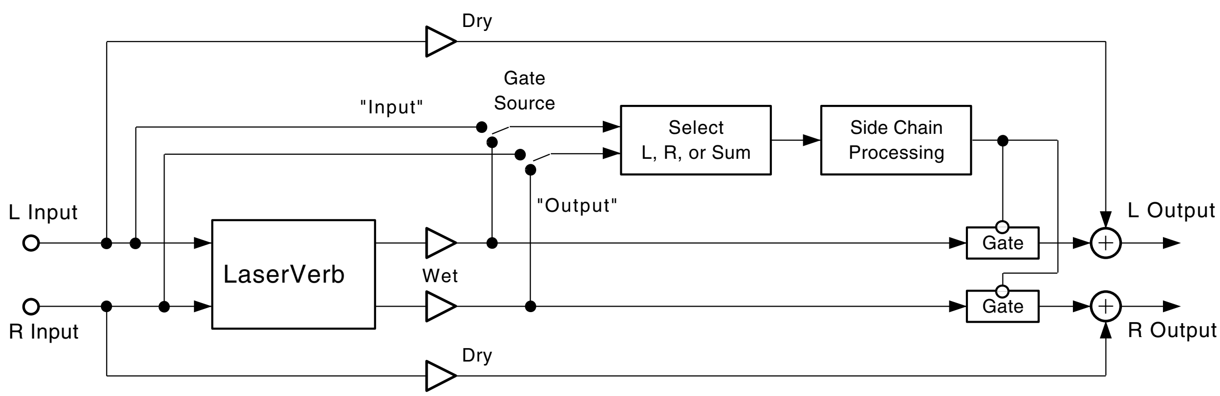


Figure 93 Signal flow of Gated LaserVerb

LaserVerb is a stereo algorithm that produces interesting sounds in the reverb decay. However, the decay often lasts longer than desired. The gate may be used to cut the output signal after the input signal drops below a threshold. You may select whether to gate the **LaserVerb** output based on the input signal level or the signal level at the output of the **LaserVerb**. In most cases the gate would be based on the input signal. When you gate on the output signal, you must wait for the **LaserVerb** tail to drop below the threshold before the gate will close. Whether you gate based on the input or the output signal strength, you can select which input or output channel to use as the gating side chain signal: left, right, or the average of the left and right magnitudes.

Parameters:

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Lvl	0 to 100 %	GateIn/Out	In or Out
Xcouple	0 to 100 %	GateSCInp	L, R, (L+R)/2
HF Damping	8 to 25088 Hz	GateSCSrc	Input or Output

Page 2

Delay Crs	0 to 5000 ms	Contour	0.0 to 100.0 %
Delay Fine	-20.0 to 20.0 ms		
Spacing	0.0 to 40.0 samp		

Page 3

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	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 The amount of reverbed and gated (wet) signal relative to unaffected (dry) signal. The gate is on the wet signal path.

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 Lite** is a stereo effect. 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.

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.

GateIn/Out Enables (**On**) or disables (**Off**) the gate. Not affected by Wet/Dry.

GateSCInp Select whether the gate side chain signal should use the left (**L**) channel, right (**R**) channel or the average magnitude of left and right channels (**(L+R)/2**) to control the gate.

GateSCSrc Select whether the gate side chain signal should be taken from the algorithm input or from the **LaserVerb** output.

Dly Coarse You can set the overall delay length from **0** to **5** seconds. 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 descending buzz and how fast it descends. 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 **0**, LaserVerb is reduced to a simple delay.

Gate Thresh The signal level in dB required to open the gate (or close the gate if Ducking 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.

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

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

GateSigDly 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.

916 Poly Pitcher

Creates pitch from pitched or non-pitched signal—twice.

PAUs: 2

Poly Pitcher is closely based on **Pitcher**, and most of the features of **Poly Pitcher** are covered in the section on **Pitcher**. **Poly Pitcher** is really just a pair of **Pitcher** algorithms (A and B) using the same inputs and summing to the same outputs. There is one set of weight parameters (Odd Wts, Pair Wts, Quartr Wts, and Half Wts), which are applied to both pitcher sections. However, the actual pitch settings for the two pitchers can be set independently. You can also set the relative level of the two pitchers with the A/B Mix parameter. One last difference from **Pitcher** is that there are separate pitch offset parameters for left and right channels for both pitchers. With separate left/right controls for the pitch offset, you can produce a greater sense of stereo separation.

Parameters:

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Odd Wts	-100 to 100 %	Quartr Wts	-100 to 100 %
Pair Wts	-100 to 100 %	Half Wts	-100 to 100 %

Page 2

A/B Mix	0 to 100 %		
Pitch A	C -1 to G 9		
PchOffs AL	-12.0 to 12.0 ST	PchOffs AR	-12.0 to 12.0 ST
Pitch B	C -1 to G 9		
PchOffs BL	-12.0 to 12.0 ST	PchOffs BR	-12.0 to 12.0 ST

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.

Odd Wts
Pair Wts
Quartr Wts
Half Wts These parameters control the exact shape of the frequency response of **Poly 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 in the section on **Pitcher**.

A/B Mix The relative amount of pitcher A and pitcher B to mix to the final output. At **0**%, only pitcher A can be heard at the output, and at **100**%, you can hear only pitcher B. **50**% produces equal amounts of both.

Pitch A, B	The fundamental pitch imposed upon the input expressed in semitone scale intervals. Pitcher A and pitcher B may be set independently.
PchOff AL **PchOff AR** **PchOff BL** **PchOff BR**	An offset from the pitch frequency in semitones. Not only are the A and B pitchers treated separately, the left and right channels have their own controls for increased stereo separation. Pitch offset may be useful as a modifiable control resembling pitch bend.

917 Frequency Offset
918 MutualFreqOffset

Single Side Band Modulation

PAUs: 2

Frequency Offset and **MutualFreqOffset** perform single side band (SSB) modulation. Essentially what this means is that every frequency component of your input sound will be offset (in frequency) or modulated by the same amount. In the **Frequency Offset** algorithm, if you have the OffsetFreq and Offs Scale parameters set to a frequency of **100 Hz**, then all frequencies in your sound will be offset up (or down) by 100 Hz. Both algorithms produce modulation both up and down and you can control the relative amount of up and down modulation with separate level and pan controls. The **Frequency Offset** algorithms are very similar to **Ring Modulator**, which is a dual side band modulator. If you set the up and down level parameters to match, the output will be quite close to the **Ring Modulator** output. Unlike **Ring Modulator** however, you can choose to listen to just the up modulation or the down modulation, and not necessarily both. In addition, you can pan the up and down modulation outputs in different directions (left or right).

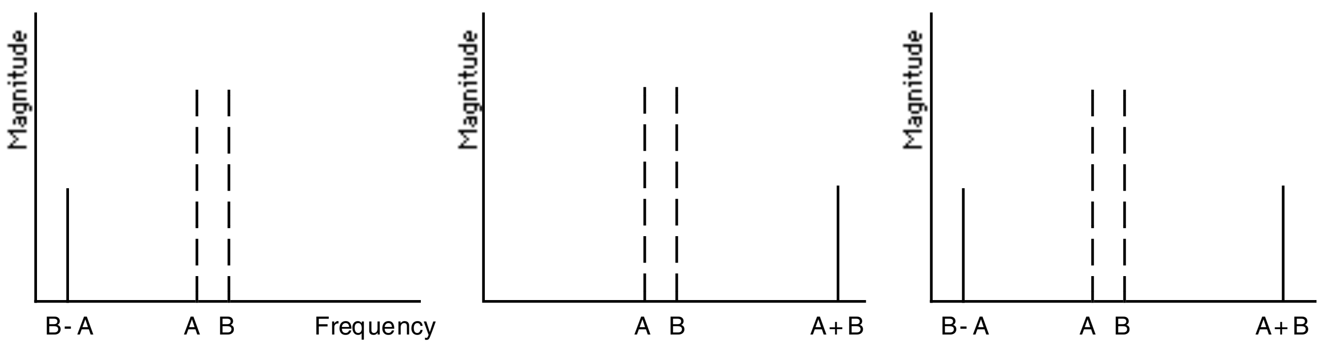


Figure 94 **Single side band modulation (frequency offset) (i) down and (ii) up. When combined (iii) we get dual side band modulation (ring modulation).**

Frequency Offset is a mono algorithm that modulates your input signal with a pure sine wave. A sine wave contains a single frequency, so your input signal will be offset in frequency by the frequency of the sine wave.

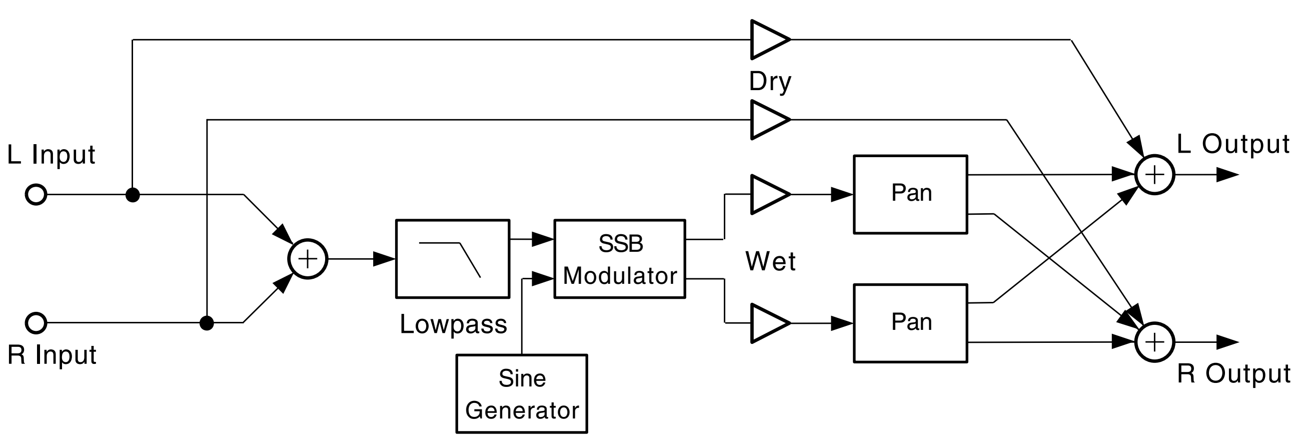


Figure 95 **Block diagram of Frequency Offset**

MutualFreqOffset modulates the two input signals (left and right) with each other. If one of the signals is a sine wave, the algorithm behaves like **Frequency Offset**. Now imagine that one of the input signals is the sum of two sine waves. Both of the two sine waves will modulate the signal on the other input. For example, if the two sine waves are at 100 Hz and 200 Hz, upward modulation of another signal at 1000 Hz will produce pitches at 1100 Hz and 1200 Hz. Obviously this is going to get very complicated to work out when the inputs are more than simple sine waves. **MutualFreqOffset** may require extra gain compensation so separate left, right input gain controls and a gain control for the final (wet) output are provided.

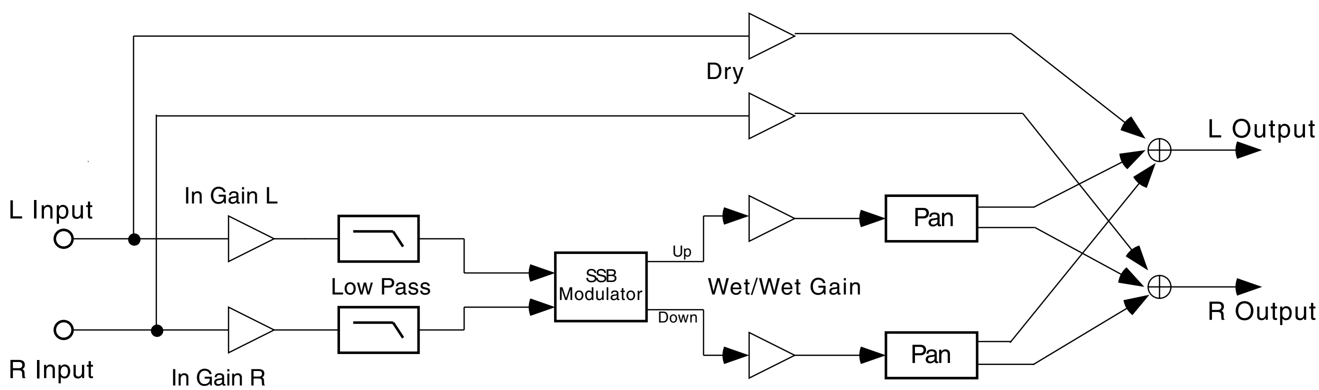


Figure 96 Block diagram of MutualFreqOffset

With downward modulation, you will hear the pitch drop as you increase the frequency of the input sound. The downward modulation is a difference (subtraction) in frequencies. If the difference drops to negative values, the frequency will start to rise again. It doesn't matter which frequency gets subtracted from the other, since the result will sound the same. For example 1000 Hz - 100 Hz = 900 Hz will produce the same pitch as 100 Hz - 1000 Hz = -900 Hz. Similarly, upward modulation is a sum of frequencies and pitch will rise as you increase the frequency of input sound. However in a digital sampled system, frequencies higher than half the sample rate (the Nyquist rate, 24 kHz in KDFX) cannot be represented. When the summed frequencies pass the Nyquist rate, the pitch starts coming back down.

Both **Frequency Offset** and **MutualFreqOffset** provide panning with width of the dry input signals directly to the output.

Parameters (Frequency Offset):

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
In Lowpass	8 to 25088 Hz		

Page 2

OffsetFreq	0.00 to 10.00 Hz		
Offs Scale	1 too25088x		
DwnOffsLvl	0 to 100 %	UpOffsLvl	0 to 100 %
DwnOffsPan	-100 to 100 %	UpOffsPan	-100 to 100 %

Parameters (MutualFreqOffset):

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
In Gain L	Off, -79.0 to 24.0 dB	Wet Gain	Off, -79.0 to 24.0 dB
In Gain R	Off, -79.0 to 24.0 dB		
InLowpassL	8 to 25088 Hz	InLowpassR	8 to 25088 Hz

Page 2

DwnOffsLvl	0 to 100 %	UpOffsLvl	0 to 100 %
DwnOffsPan	-100 to 100 %	UpOffsPan	-100 to 100 %

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

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

In Lowpass A first-order lowpass filter is provided to reduce the bandwidth of the input signal. Considering the many new frequency components that will be created, the lowpass filter may help tame the sound. **MutualFreqOffset** has separate controls for left and right input channels.

OffsetFreq **Frequency Offset** algorithm only. The frequency when multiplied with Offs Scale which is the modulation frequency. The offset or modulation frequency is the frequency in Hz which is added to and/or subtract from all the frequencies of the input signal.

Offs Scale **Frequency Offset** algorithm only. A scale factor which is multiplied with the OffsetFreq parameter to produce the offset or modulation frequency.

In Gain L/R **MutualFreqOffset** algorithm only. Two independent gain controls (left and right) to adjust the amplitude of the input signals. (See Wet Gain.)

Wet Gain The gain or amplitude of the modulated (wet) signal. The Wet Gain parameter and the In Gain L/R parameters are for the **MutualFreqOffset** algorithm which produces an output based on multiplying the left and right inputs. This is very different from adding signals, and controlling levels can be tricky. Ideally you would set the input gains and the wet gain so that the signal level remains flat when you adjust Wet/Dry while ensuring you hear no internal clipping. Use Out Gain for overall level control.

DwnOffsLvl The level of the down modulated signal. Negative values polarity invert the signal.

UpOffsLvl The level of the up modulated signal. Negative values polarity invert the signal.

DwnOffsPan The down modulated signal may be panned to the left or right algorithm outputs. **-100%** sends the signal to the left output and **100%** sends the signal to the right output.

UpOffsPan The up modulated signal may be panned to the left or right algorithm outputs. **-100%** sends the signal to the left output and **100%** sends the signal to the right output.

919 WackedPitchLFO

An LFO based pitch shifter.

PAUs: 3

Okay, it ain't pretty, but **WackedPitchLFO** uses LFO modulated delay lines with cross fades to produce a shift of signal pitch. You can set the amount of shift in coarse steps of semitones or fine steps of cents (hundredths of a semitone). This shifter works using the same concepts used to detune a sound in a chorus algorithm. In a chorus algorithm, an LFO is used to change the length of a delay line. By smoothly changing a delay line length from long to short to long, the signal is effectively resampled at a new rate causing the pitch to rise and fall. In the **WackedPitchLFO** algorithm, the signal level is made to rise and fall in time with the delay line movement so that we only hear signal from the delay line when the pitch is rising (or falling). By overlapping and adding several delay taps moved by several LFOs, we can then produce a relatively smooth pitch shifted signal.

Relatively.

It is possible for sounds coming out of the delay lines to be out of phase, which means that a certain amount of cancellation can occur. The result sounds like there is a certain amount of tremolo in the pitch shifted signal. The depth of the tremolo will depend on the pitch of the signal, the rate of the LFO and the amount of pitch shifting—it will be different for every pitch. The rate of the tremolo is the rate of the LFO. At higher rates the tremolo can be objectionable. At slow LFO rates, the pitch shifting is quite clean, though you will hear some flanging. However longer delay line lengths are needed at slower LFO rates for a given amount of pitch shift. The delays can get quite long, and it is possible to run out of available delay (in which case you will get less pitch shift than you request). The trade-off is tremolo for delay. Higher frequency signals will sound better when pitch shifted than lower frequency signals. Increasing the amount of pitch shift will increase both the amount of tremolo and the amount of delay.

You can introduce feedback in **WackedPitchLFO**. When you do, the signal can be made to continuously rise (or fall) as it repeatedly passes through the feedback loop.

The pitch shifter is based on delay lines. Changing the amount of pitch shift will produce large jumps in delay line lengths, and you will hear the jumps as clicks if you are playing a sound while changing the shift amount. For this reason, the shift amount parameters will not work well as modifiable parameters on an FXMOD page.

Parameters:

Page 1

Wet/Dry	-100 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Feedback	0 to 100 %	Highpass	8 to 25088 Hz
LFO Rate	0.01 to 10.00 Hz	Shift Crs	-24 to 24 ST
Lowpass	8 to 25088 Hz	Shift Fine	-100 to 100 ct

Wet/Dry The relative amount of input signal and pitch shifted 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 By introducing feedback, the pitch can be made to continually rise or fall as the signal makes successive passes through the pitch shifter.

LFO Rate The frequency of the LFOs that drive the pitch shifter. The pitch shifter produces a certain amount of tremolo that will oscillate based on this rate. However reducing the rate will increase the delay lengths needed by the pitch shifter.

Shift Crs A coarse adjust to the pitch shift amount from -24 to +24 semitones. The algorithm performs best when the amount of pitch shift is small.

Shift Fine A fine adjust to the pitch shift amount from -100 to +100 cents (hundredths of a semitone).

Lowpass A lowpass filter in the algorithm feedback loop. Use the lowpass to tame some of the higher frequency artifacts. This is especially important when using feedback.

Highpass A highpass filter in the algorithm feedback loop. Use the highpass to tame the lower frequencies when using feedback.

920 Chaos!

Fun with chaos and instability

PAUs: 2

The moment you scroll to the **Chaos!** algorithm, you will discover it is wildly unstable. **Chaos!** is a delay feedback algorithm which includes lots of gain with distortion plus plenty of filters tweaking the sound. Modifying the parameters will often cause the algorithm to jump from one chaotic instability state to another, often unpredictably. For the most part **Chaos!** howls and resonates on its own, and while an input signal can affect the output, the effect of the input signal on the output is usually small. When self-resonating, the sound you can get can be very strange. It is particularly interesting if you keep modifying the parameters. What do you use this effect for? Well, that's the creative challenge!

You should be very careful with the Out Gain or Drive Cut settings with **Chaos!** If you start the algorithm in a stable state (not self resonating) and start increasing gains (in the distortion drive or filters), the output level can build. The feedback can be every bit as unpleasant as putting a microphone next to a loudspeaker! (There's an application: simulating PA system feedback!)

Let's take a closer look at **Chaos!**

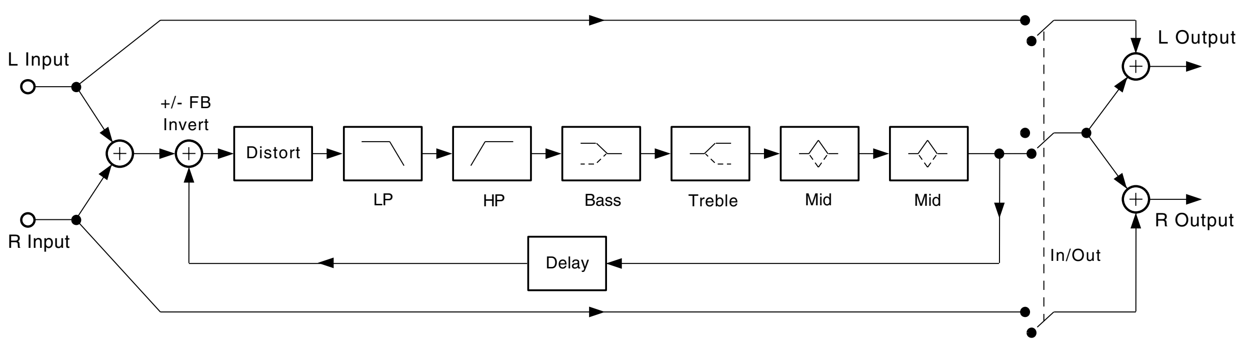


Figure 97 Chaos!

As advertised, **Chaos!** is a feedback loop with delay, distortion and lots of filters. Most of the effects in KDFX carefully manage levels on feedback loops to prevent instability. In a digital system, uncontrolled instability will usually rapidly enter digital clipping with full scale signal output. Very nasty. **Chaos!** also keeps a lid on levels, preventing digital clipping but allowing instability. You will still need to cut back on Out Gain (or Drive Cut) to bring the signal down to reasonable levels.

The distortion drive when turned up, will push **Chaos!** into instability unless Drive Cut is used to hold the level down. As the sound starts becoming unstable, your input signal will still have a strong effect on the output. As more and more drive is applied, the self-resonance dominates the output.

The delay length is expressed as a frequency where the length of the delay in seconds is 1/frequency. Why do this? A short delay line with a lot of feedback will resonate at a frequency of 1/length of the delay. It is the resonant behavior of **Chaos!** which is particularly interesting, which make the delay more naturally expressed as a frequency. Not only will the delay resonate at its natural frequency (1/length), but you may also hear many overtones (or harmonics). There is a switch to invert the feedback (FB Invert). When set to In, FB Invert will cause the natural frequency and its harmonics to be suppressed while frequencies between the harmonics now resonate. In this case the frequency one octave down and its odd harmonics are resonating.

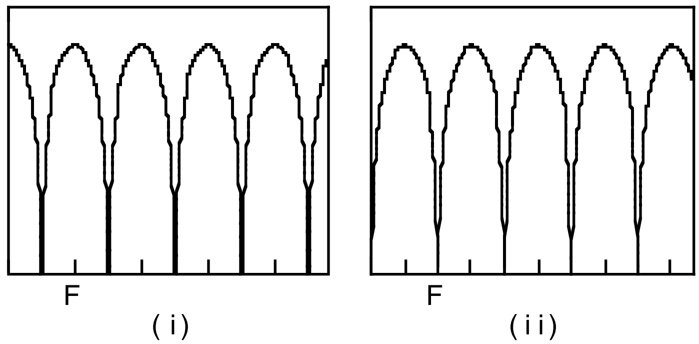


Figure 98 **Resonating frequencies with FB Invert set to (i) Out and set to (ii) In.**

In addition to the distortion warmth filter, there are six filters built into the delay line loop: a highpass, a lowpass, a treble and a bass shelf, and two parametric midrange filters. Boosting the shelves or mids increases the strength of instability at the boosted frequencies. Since overall level is controlled, the net effect is to reduce the level of the other frequencies. Using filters to cut frequencies is similar, but with cut it is possible to remove so much signal that the algorithm drops into stability and stops self-resonating.

The individual elements of **Chaos!** (filters and so forth) are fairly basic, and you may understand them well. When put together as the **Chaos!** algorithm, the interactions become very complex and many of the old rules don't seem to apply. Keep plugging at it.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Drive	0 to 96 dB	FB Invert	In or Out
Drive Cut	Off, -79.0 to 0.0 dB	Dly FreqC	8 to 25088 Hz
Warmth	8 to 25088 Hz	Dly FreqF	-100 to 100 ct

Page 2

Highpass	8 to 25088 Hz	Lowpass	8 to 25088 Hz
Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz
Mid1 Gain	-79.0 to 24.0 dB	Mid2 Gain	-79.0 to 24.0 dB
Mid1 Freq	8 to 25088 Hz	Mid2 Freq	8 to 25088 Hz
Mid1 Width	0.010 to 5.000 oct	Mid2 Width	0.010 to 5.000 oct

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. The output gain is outside and after the feedback loop.

Drive Sets how high the distortion is to be driven. The distortion and its drive gain are inside the feedback loop.

Drive Cut Reduces the signal level after the distortion. By reducing the signal level after the distortion, **Chaos!** can be returned to stability while still producing a lot of distortion. Drive Cut is also inside the feedback loop.

Warmth Warmth affects the character of the distortion. Warmth reduces (at low settings) the higher frequency distortion components without making the overall signal dull.

Dly FreqC The feedback signal path includes a short delay line which will tend to resonate at a frequency of 1/length of the delay. The delay length is therefore expressed as the resonant frequency. Note that all the filters in the feedback loop also add delay, so with more filtering, the resonance tuning will drift flat.

Dly FreqF The resonant frequency of the feedback delay line can be tuned sharp or flat in one cent (hundredths of a semitone) increments.

FB Invert The feedback signal can be inverted (subtracted instead of added) so that instead of resonance at the specified frequency and its harmonics, the resonance occurs between those frequencies. This is like setting resonance one octave lower, but using only the odd harmonics.

Highpass The highpass filter removes frequencies below the specified cut-off frequency. The filter is first order, cutting signal level at 6 dB per octave of frequency. When set to the lowest frequency, the filter is performing very little cut of the low frequencies. When **Chaos!** is self-resonating, turning up the highpass frequency will cause high frequencies to be emphasized.

Lowpass The lowpass filter removes frequencies above the specified cut-off frequency. The filter is first order, cutting signal level at 6 dB per octave of frequency. When set to the highest frequency, the filter is performing very little cut of the high frequencies. When **Chaos!** is self-resonating, turning down the lowpass frequency will cause low frequencies to be emphasized.

Bass Gain The amount of boost or cut in decibels to apply to the bass shelf filter inside the feedback loop. Boost will emphasize frequencies below the filter frequency, while cut will emphasize frequencies above the filter frequency.

Bass Freq The frequency in Hz below which the bass shelf filter performs boost or cut.

Treb Gain The amount of boost or cut in decibels to apply to the treble shelf filter inside the feedback loop. Boost will emphasize frequencies above the filter frequency, while cut will emphasize frequencies below the filter frequency.

Treb Freq The frequency in Hz above which the treble shelf filter performs boost or cut.

Mid*n* Gain The amount of boost or cut in decibels to apply to the midrange parametric filter n (1 or 2) inside the feedback loop. Boost will emphasize the specified filter frequency while cut will emphasize all other frequencies.

Mid*n* Freq The frequency in Hz at which the midrange parametric filter n (1 or 2) performs boost or cut.

Mid*n* Width The width of the frequency band in octaves of the midrange parametric filter n (1 or 2). When the filter is set for boost, a narrow band (low settings) will cause the resonating output to approach a pure tone more rapidly.

948 Band Compress

Stereo algorithm to compress a single frequency band

PAUs: 3

Band Compress is in most respects identical to **SoftKneeCompress**. However, **Band Compress** compresses only on a single band of frequencies. Frequency band selection is based on a parametric filter. You control the filter center frequency and bandwidth. The compressor controls the filter gain.

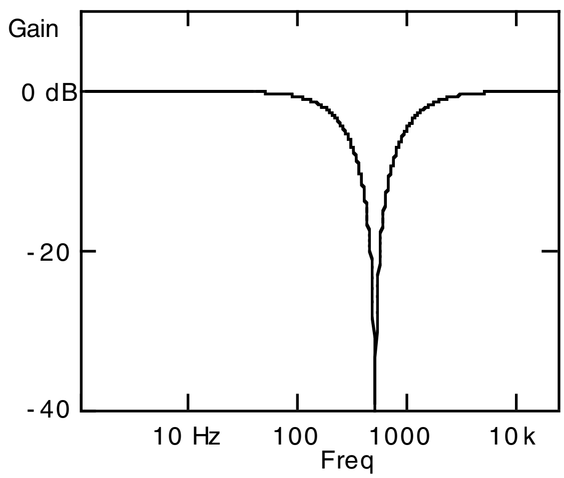


Figure 99 Band Compress filtering at full compression

The compressor reduces the signal level when the signal level 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.

With **Band Compress**, the side chain processing acts only on a specified band of frequencies. At the input of the side chain is a bandpass filter which passes only the frequency band of interest. The side chain output then controls the gain of a band cut filter acting on the main signal path. The bandpass and band cut filters both are set to the same center frequency and bandwidth which you control. The depth of the band cut filter is of course set by the compressor side chain processing.

You can select which channel, left (**L**), right (**R**) or the maximum amplitude of the two (**L & R**) is used to control the compression (side chain processing) with the SC Input parameter. You can also select which channel is actually compressed, again left (**L**), right (**R**) or both (**L & R**) using the ComprsChan parameter.

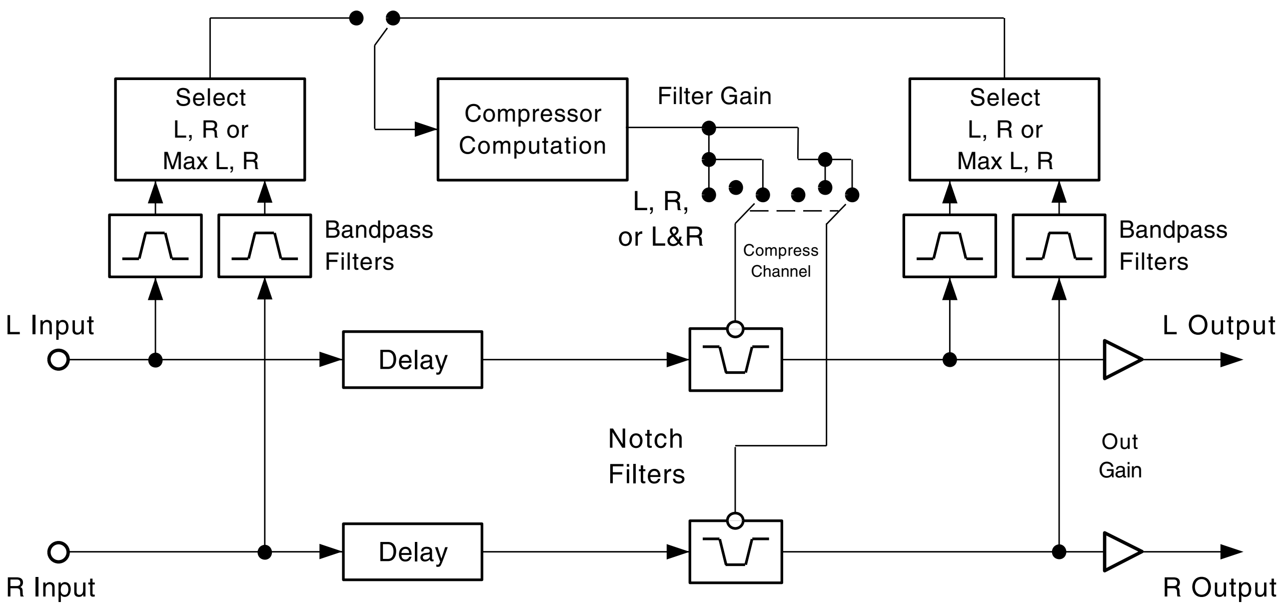


Figure 100 Band Compress block diagram

The soft-knee compressor is used which has a more gradual transition from compressed to unity gain.

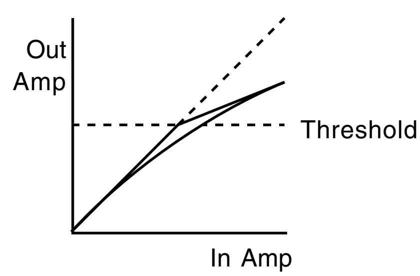


Figure 101 Soft-Knee compression characteristic

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 overshoot 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 behavior, 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.

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
Band FreqC	8 to 25088 Hz	FdbkCmprs	In or Out
Band FreqF	-100 to 100 ct	SC Input	L, R, L & R
Band Width	0.010 to 5.000 oct	ComprsChan	L, R, L & R

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 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		
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

In/Out When set to **In** the compressor is active; when set to **Out** the compressor 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**.

Band FreqC The coarse control for the center frequency of the filter band to be compressed. Only signal components centered in the band will be compressed.

Band FreqF The fine control for the center frequency of the filter band to be compressed. Only signal components centered in the band will be compressed.

Band Width The width of the frequency band to be compressed in octaves. Small values compress a very narrow range of frequencies. Large values compress a broad range of frequencies.

SC Input Select the input source channel for side-chain processing—left (**L**), right (**R**) or both (**L & R**). When set to **L & R**, the maximum amplitude is used.

ComprsChan Select which input channel will receive compression processing—left, right or both. If you select left or right, the opposite channel will pass through unaffected.

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 compressor'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 predelay). This allows the compression to appear to take effect just before the signal actually rises.

Ratio The compression ratio in effect above the compression threshold. High ratios are highly compressed; low ratios are moderately compressed.

Threshold The compression 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.

949 CompressDualTime

Compression with 2 release time constants

PAUs: 2

CompressDualTime is a basic compressor with two different release rates, which change from one rate to another as the compression gain reduction crosses a threshold set by the Rel Thres (release threshold) parameter. Except for the additional release rate and release threshold parameters, this compressor is like **SoftKneeCompress**.

Like a normal compressor, this compressor reduces the gain of a signal as the signal increases above the compression threshold set with CompThres parameter. 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 side chain input signal (left, right or larger of left and right magnitudes). 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.

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 overshoot 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.

CompressDualTime has a special way of handling the release time. There are in fact two release times: Rel Time A and Rel Time B. There is also a release threshold (Rel Thres) parameter. While the amount of compression gain reduction exceeds the release threshold, the release time will be determined by Rel Time A. As soon as the amount of reduction is less than the threshold, the release time changes to Rel Time B.

You can set the release times so that the release is initially very fast, and then let it slow down when it crosses the release threshold. Likewise, you can have a very slow release, which suddenly speeds up as reduction reaches the release threshold. Be sure to watch the reduction meter to follow what is happening as you change the parameters.

You can select which channel, left (**L**), right (**R**) or the maximum amplitude of the two (**L & R**) is used to control the compression (side chain processing) with the SC Input parameter. You can also select which channel is actually compressed, again left (**L**), right (**R**) or both (**L & R**) using the ComprsChan parameter.

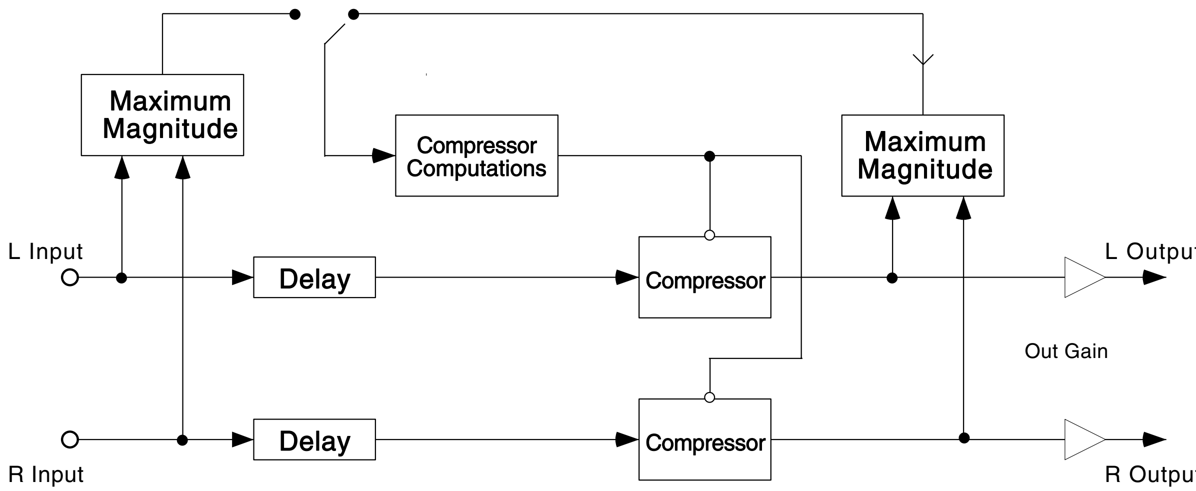


Figure 102 Opto Compress

The soft-knee compressor is used which has a more gradual transition from compressed to unity gain.

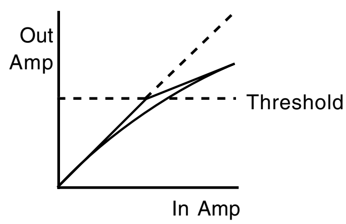


Figure 103 Soft-Knee compression characteristic

For typical compressor behavior, 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.

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	SC Input	L, R, L & R
Signal Dly	0.0 to 25.0 ms	ComprsChan	L, R, L & R

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1:1.0 to 1:17.0
Rel Time A	0 to 3000 ms	Comp Thres	-79.0 to 0.0 dB
Rel Time B	0 to 3000 ms	Rel Thres	-79.0 to 0.0 dB
SmthTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12	8 6 4 2	0

In/Out When set to **In** the compressor is active; when set to **Out** the compressor 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**.

SC Input Select the input source channel for side-chain processing—left (**L**), right (**R**) or both (**L & R**). When set to **L & R**, the maximum amplitude is used.

ComprsChan Select which input channel will receive compression processing—left, right or both. If you select left or right, the opposite channel will pass through unaffected.

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 A The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold. This release time is active while the signal is reduced by more than the release threshold setting.

Rel Time B The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold. This release time is active while the signal is reduced by less than the release threshold setting.

Rel Thres When the signal is reduced by more than this release threshold, the release time is set by Rel Time A. Otherwise the release time is set by Rel Time B.

SmthTime A lowpass filter in the control signal path. It is intended to smooth the output of the compressor'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 predelay). This allows the compression to appear to take effect just before the signal actually rises.

Ratio	The compression ratio in effect above the compression threshold. High ratios are highly compressed; low ratios are moderately compressed.
Comp Thres	The compression 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.

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.

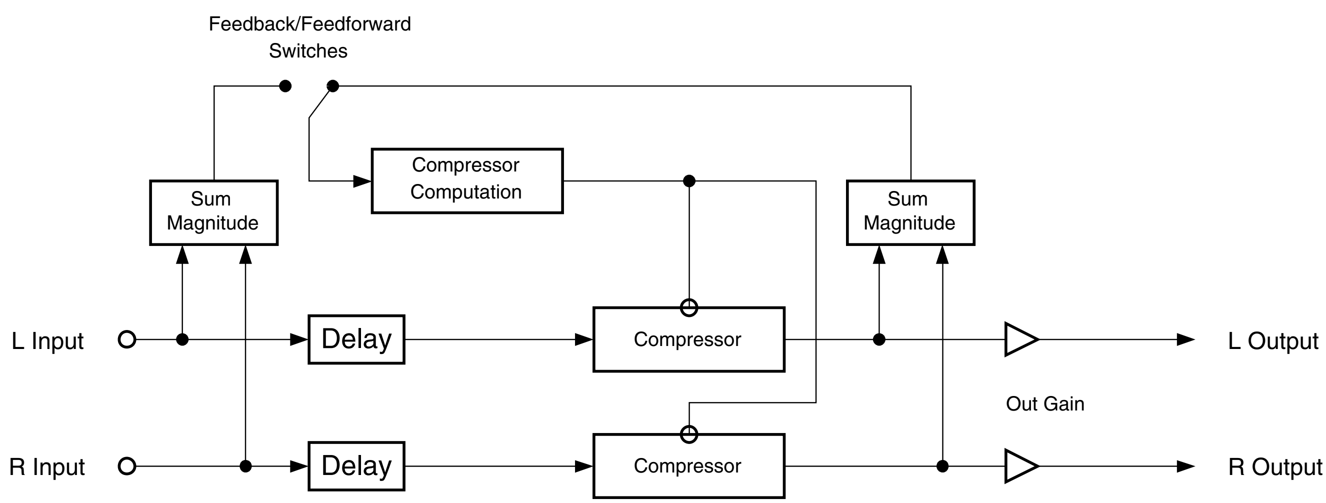


Figure 10-104 Compressor

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.

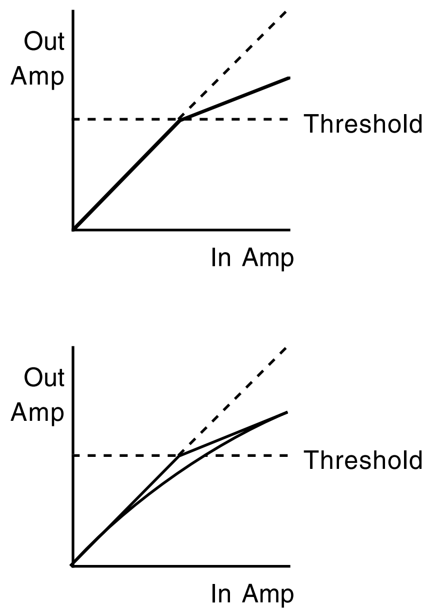


Figure 10-105 Hard- and Soft-Knee Compression Characteristics

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 behaviorbehavior, 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.

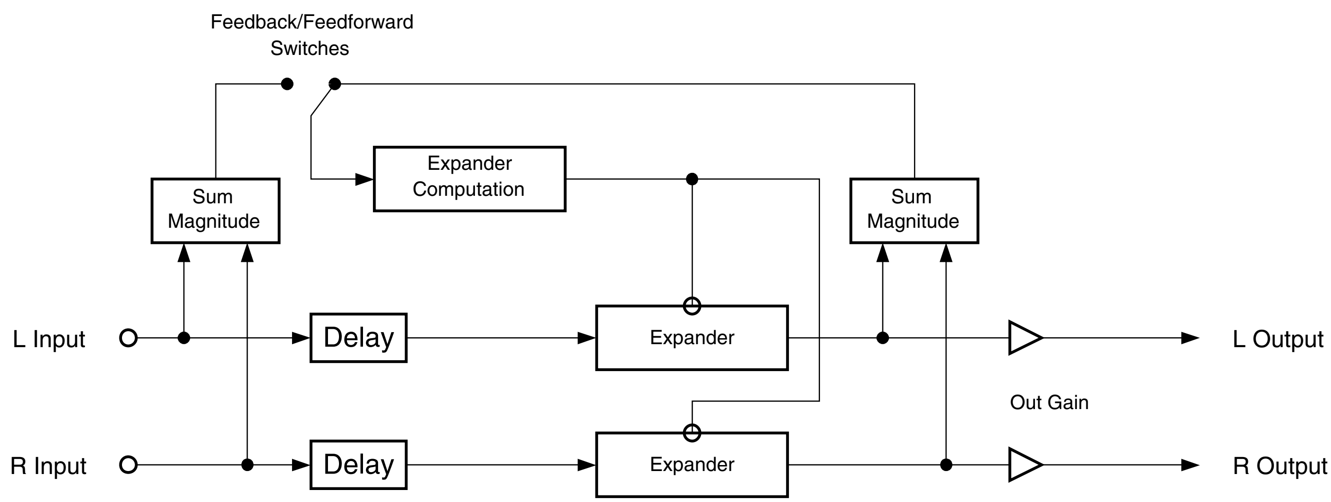


Figure 10-106 Expander

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.

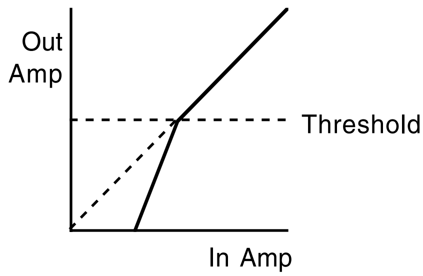


Figure 10-107 Expansion Transfer Characteristic

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.

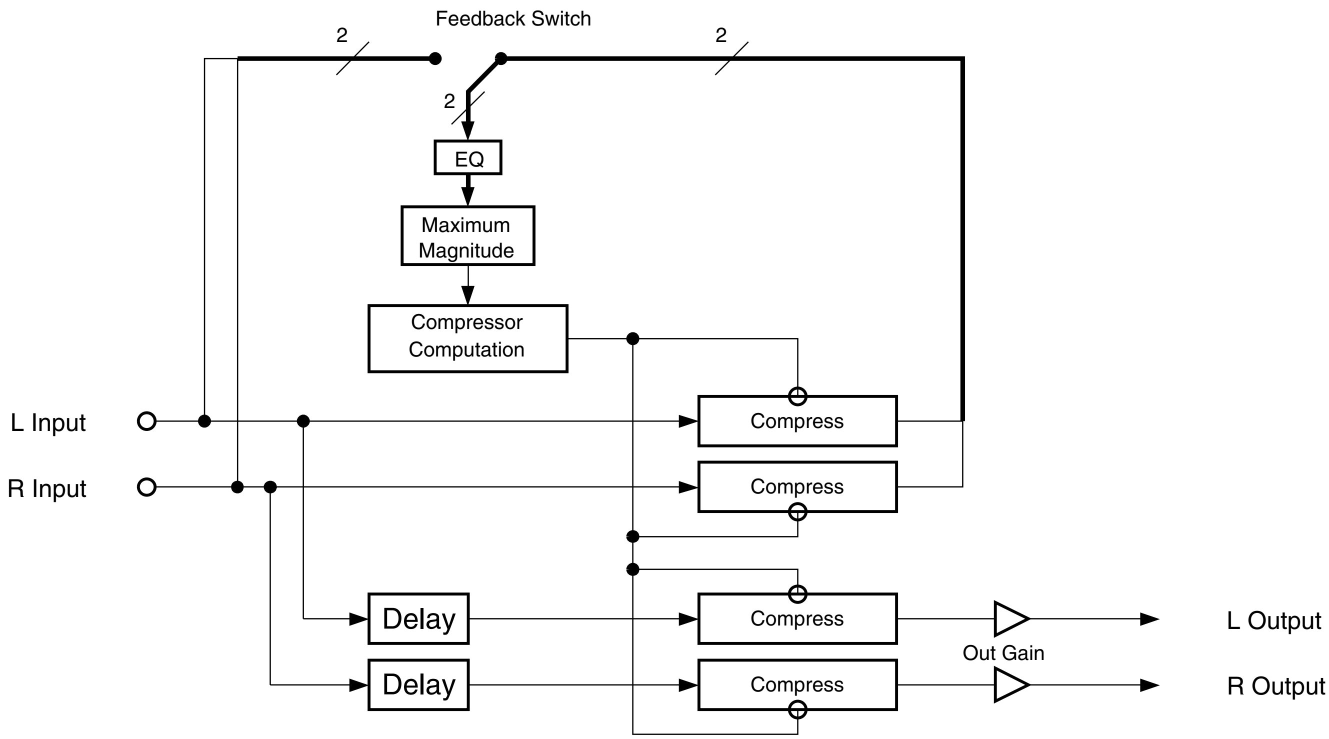


Figure 10-108 Compressor with side chain equalization.

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.

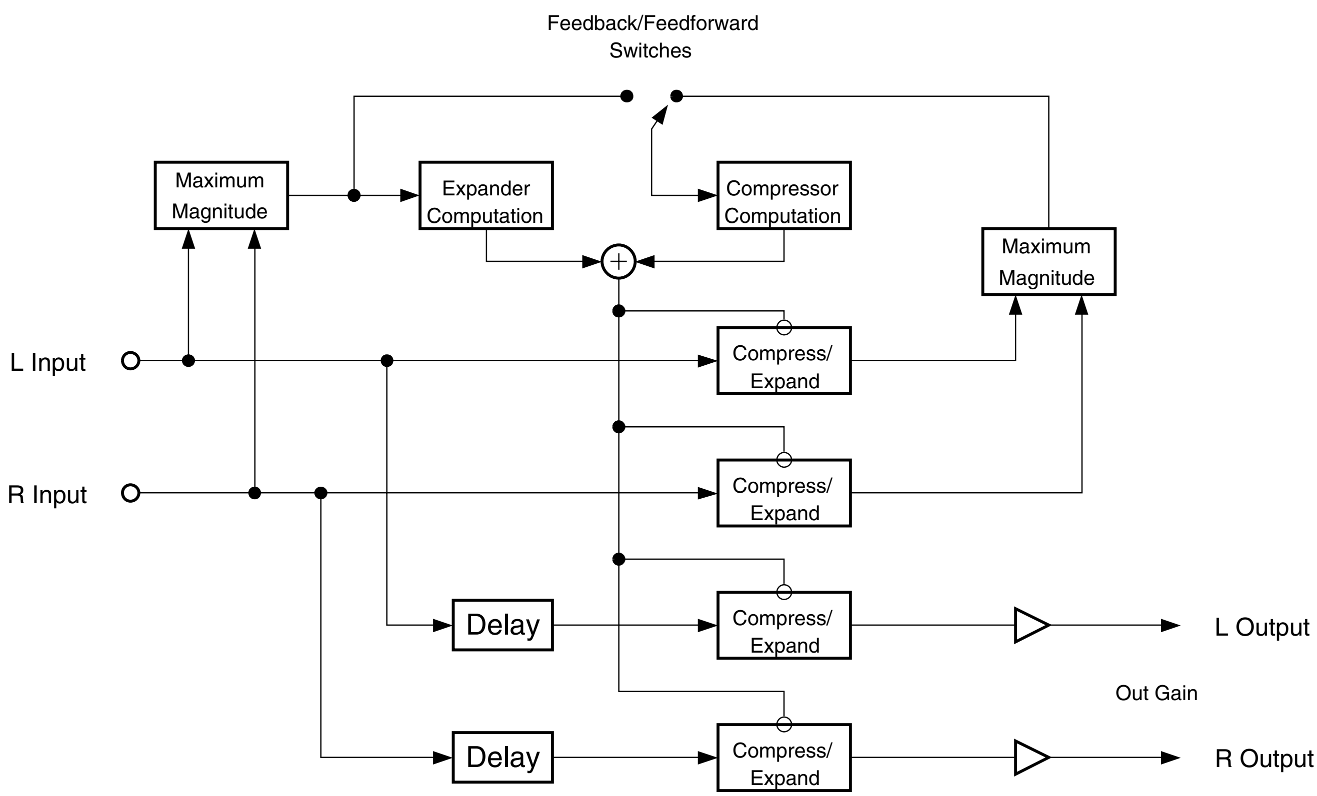


Figure 10-109 Compressor/Expander (optional EQ not shown)

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 behavior, 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).

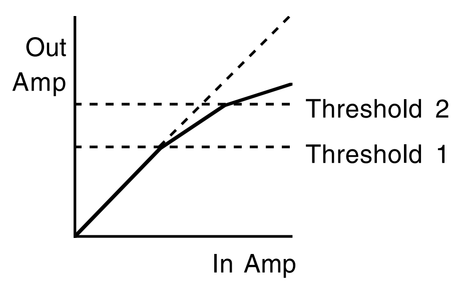


Figure 10-110 Two Segment Compression Characteristic

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.

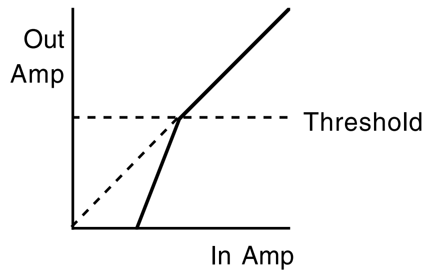


Figure 10-111 Expansion Transfer Characteristic

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		

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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		

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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.

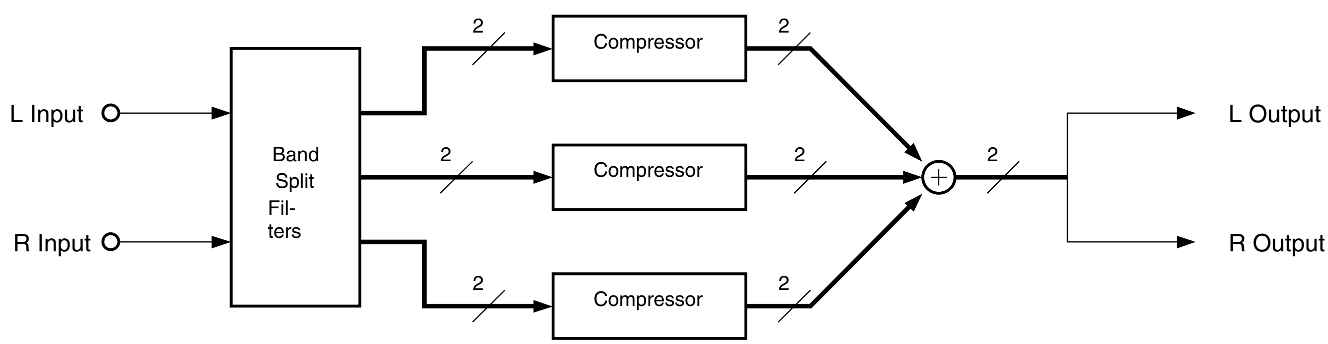

Figure 10-112 Band Compressor

In the soft-knee compressor there is a gradual transition from compressed to unity gain.

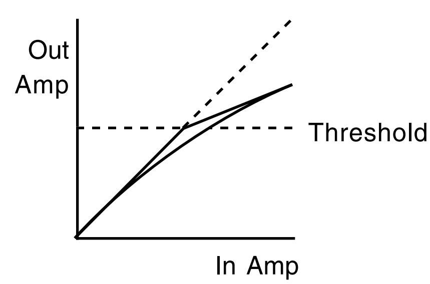

Figure 10-113 Soft-Knee Compression Characteristics

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 behavior, 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.

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

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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

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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

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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.

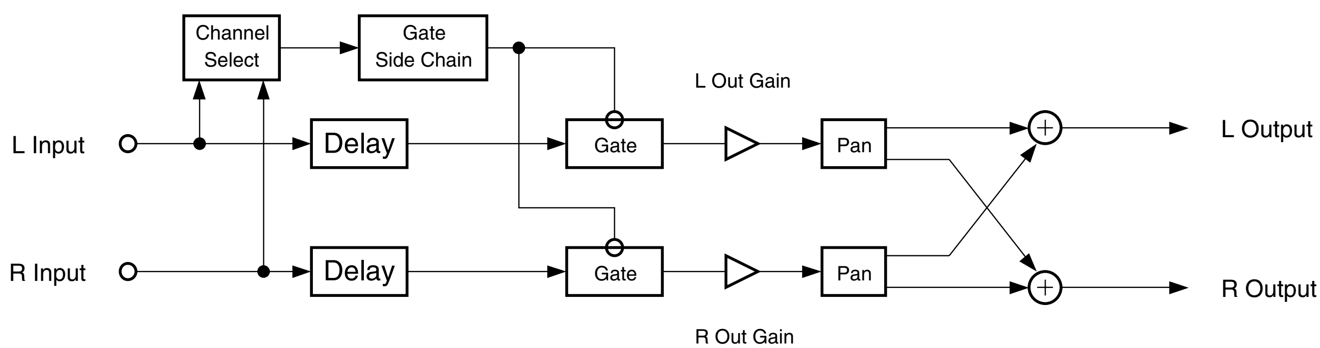


Figure 10-114 Gate

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.

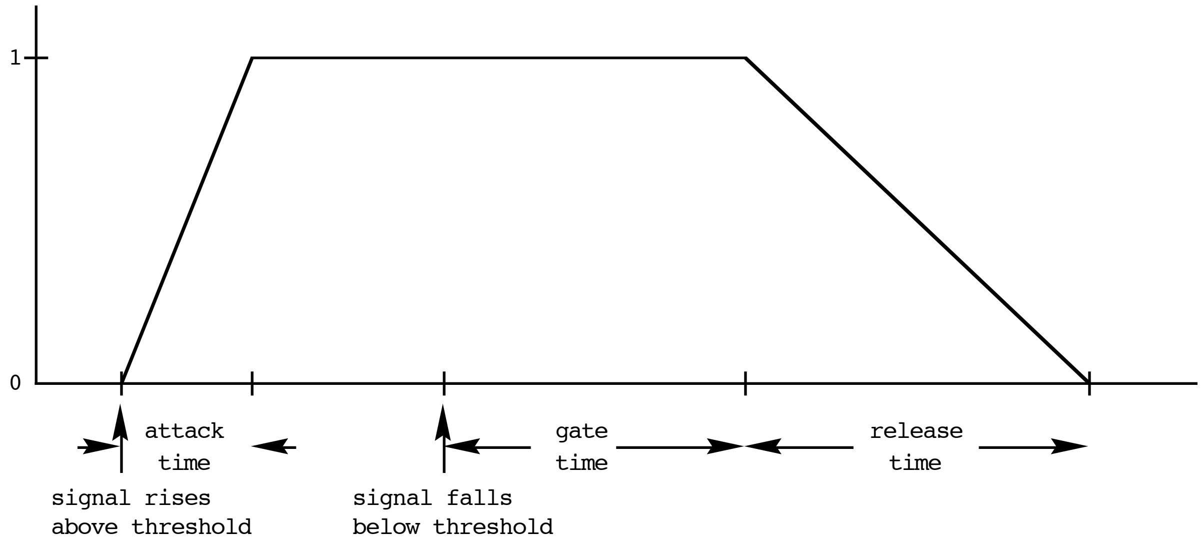


Figure 10-115 Signal envelope for Gate and Super Gate when Retrigger is "On"

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.

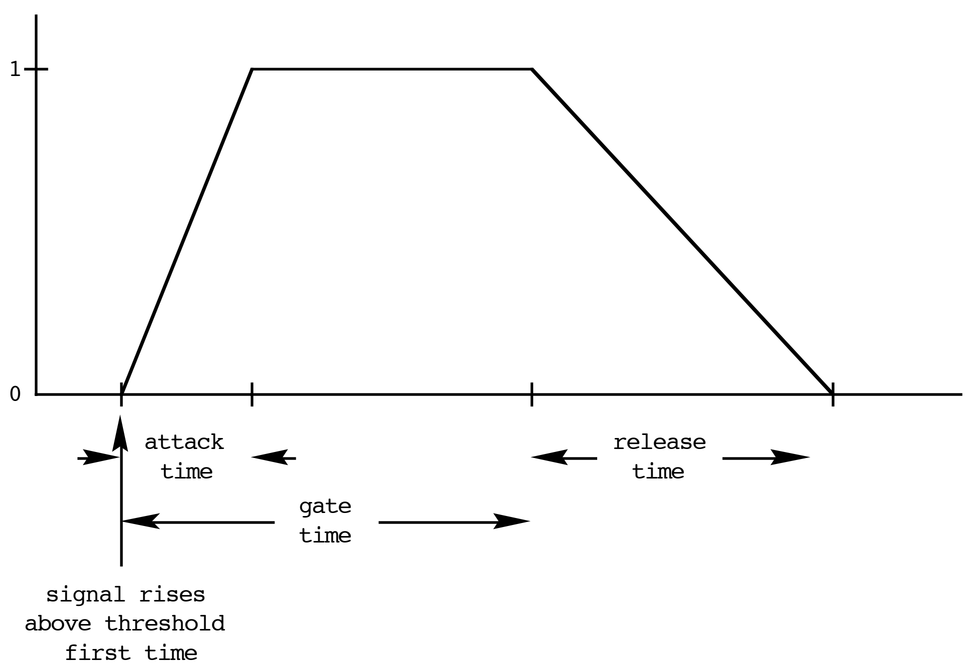


Figure 10-116 Super Gate signal envelope when Retrigger is "Off"

If Ducking is turned on, then the behavior 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.

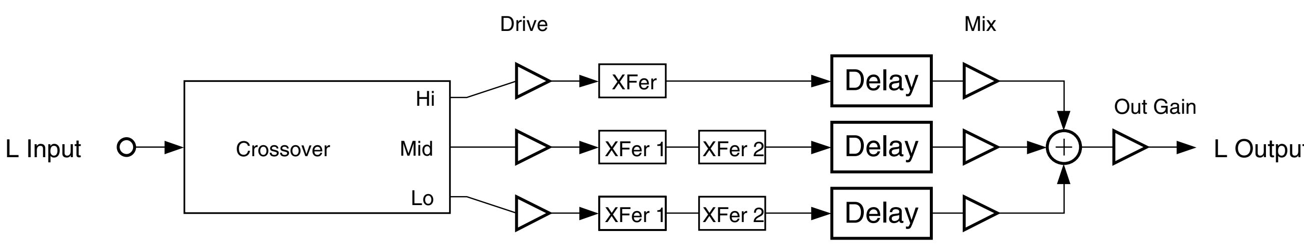


Figure 10-117 One channel of 3 Band Enhancer

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.

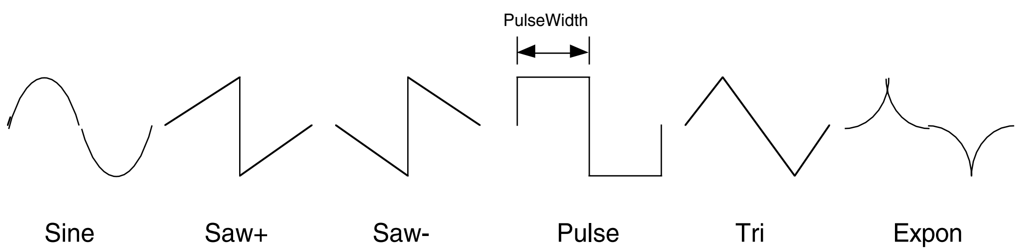


Figure 10-118 LFO Shapes available for Tremolo and Tremolo BPM

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.

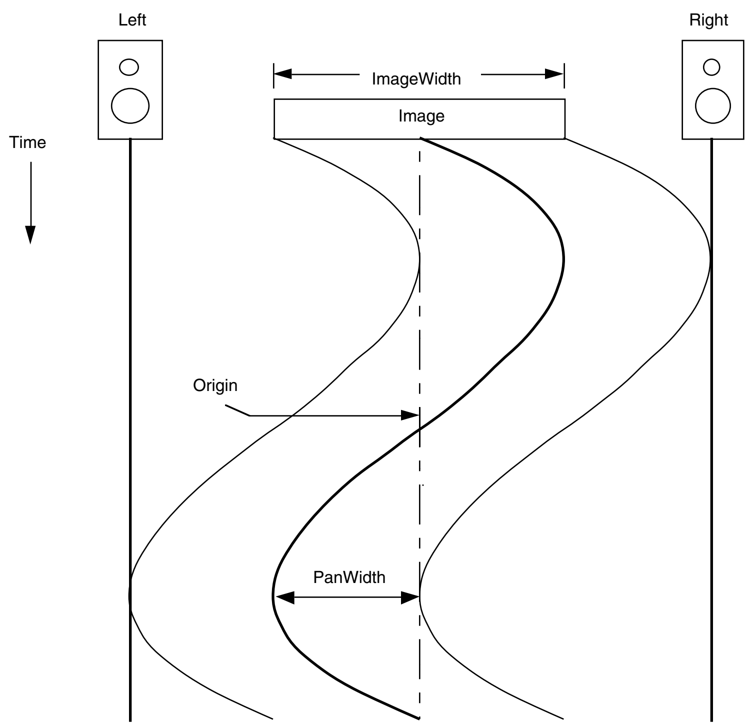


Figure 10-119 Stereo Autopanning

In Figure 10-119, ImageWidth is set to 50%, LFO Shape is set to Sine, Origin is set to 0%, and PanWidth is set to 100%

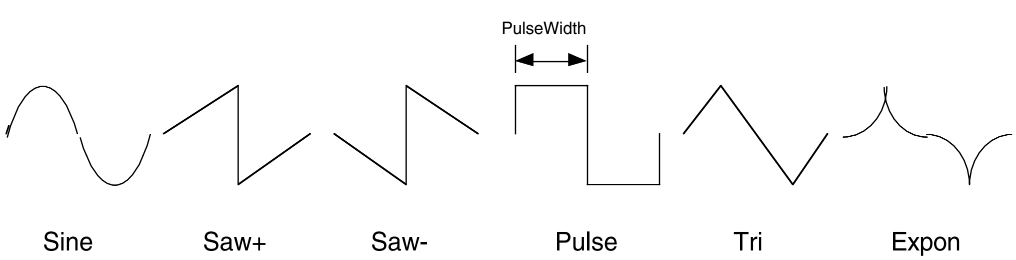


Figure 10-120 LFO Shapes available for AutoPanner

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.

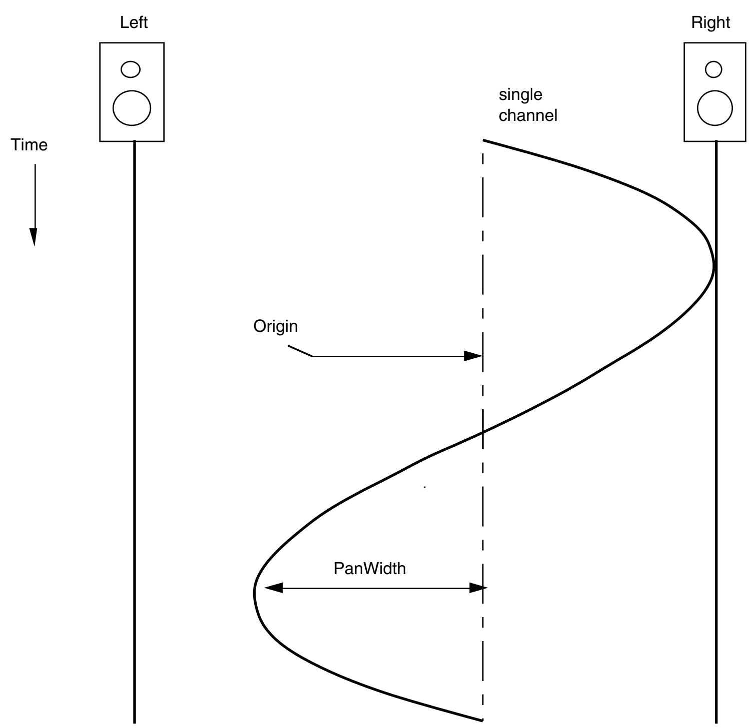


Figure 10-121 Mono autopanning

In Figure 10-121, LFO Shape is set to Sine, Origin is set to 15%, and PanWidth is set to 100%

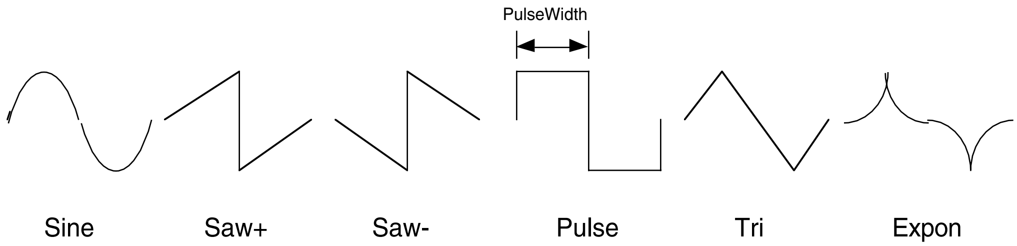


Figure 10-122 LFO Shapes available for Dual AutoPanner

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.

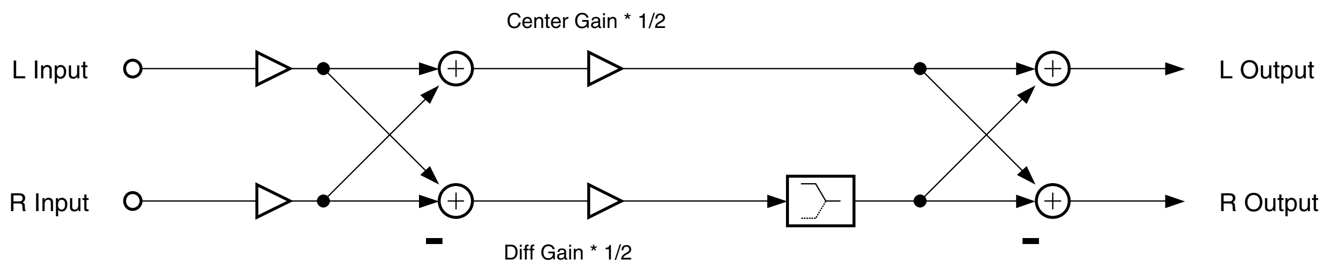


Figure 10-123 Block diagram of Stereo Image algorithm

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.

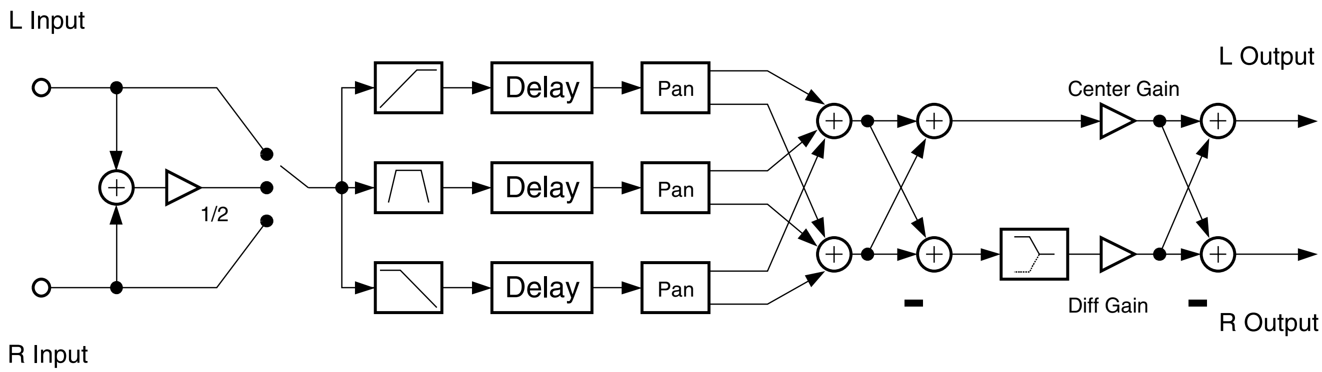


Figure 10-124 Block diagram of Mono -> Stereo effect.

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).

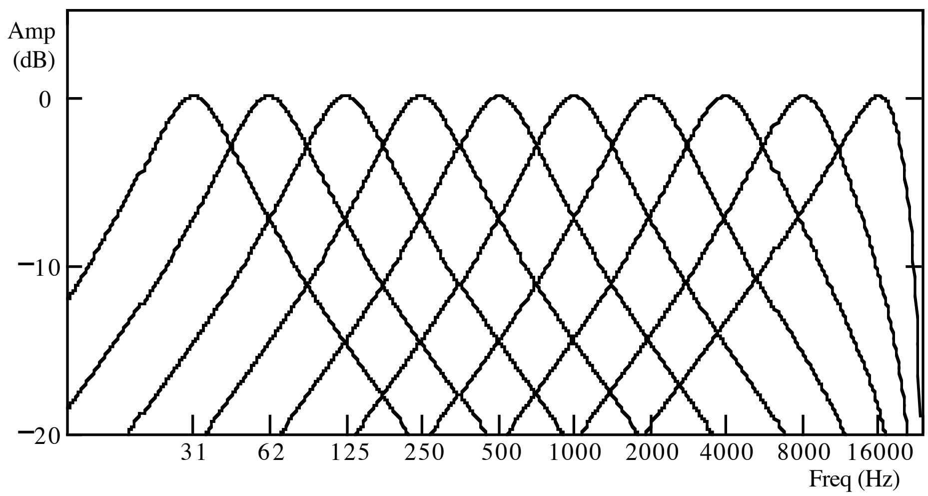


Figure 10-125 Filter Response of Each Bandpass Filter

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.

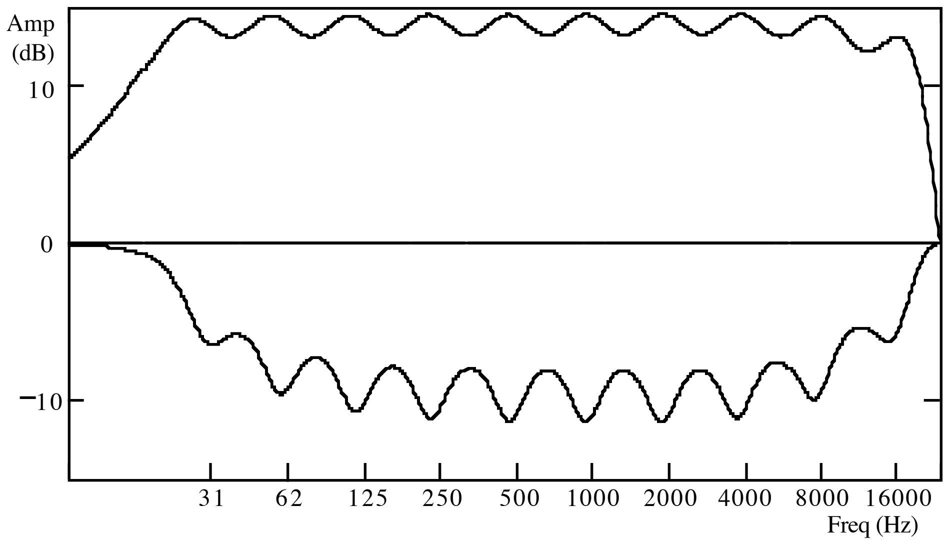


Figure 10-126 Overall Response with All Gains Set to +12 dB, 0 dB and -6 dB

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.

971 3 Band EQ

Bass and treble shelving filter and parametric EQs

PAUs: 1

This algorithm is a multi-band equalizers with parametric EQ and bass and treble tone controls. You can control 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 small **3 Band EQ** does not provide control of the bandwidth for the parametric Mid filter.

The algorithm **3 Band EQ** is stereo, meaning the parameters for the left and right channels are ganged—the parameters have the same effect on both channels.

Parameters:

Page 1 (3 Band EQ)

In/Out	In or Out	Mid Gain	-79.0 to 24.0 dB
Out Gain	-79.0 to 24.0 dB	Mid Freq	8 to 25088 Hz
Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz

In/Out When set to **In** the tone controls are active; when set to **Out** the tone controls are bypassed.

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

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 filter 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 filter in intervals of one semitone.

Mid 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 Freq The center frequency of the EQ in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

972 HF Stimulate 1

High-frequency stimulator

PAUs: 1

The high-frequency stimulator algorithm is closely based on the V.A.S.T. High Frequency Stimulator DSP function, and the manual description is repeated here (edited for KDFX specifics).

The overall effect of a high-frequency stimulator is to boost the high frequency partials of the signal, and depending on the settings of the parameters, it can add high-frequency partials to the signal as well. It's useful for building sounds that cut through the mix, and have a bright crisp nature.

There's more to the high-frequency stimulator than meets the eye. It works like this: the signal is run through a highpass filter, then through a distortion function, then through a second highpass filter. Finally. it's mixed with the original signal after passing through the final Stim Gain level control of the algorithm.

The **HF Stimulate 1** algorithm is a close copy of the V.A.S.T. DSP function, giving control of the first highpass filter frequency, the distortion drive and the amplitude of the result (Stim Gain). As a bonus, the distortion curve can also be adjusted.

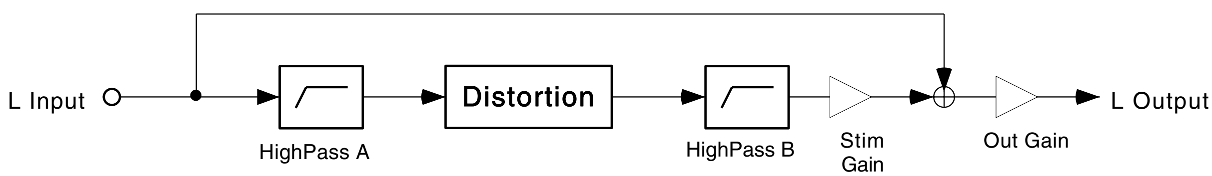


Figure 127 One channel of high-frequency stimulation

Parameters:

Page 1

Stim Gain	Off, -79.0 to 24.0 dB	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	-79.0 to 48.0 dB	Highpass	8 to 25088 Hz
Dist Curve	0 to 127%		

Stim Gain The gain of the high frequency stimulated signal applied prior to being added to the original input signal.

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

Dist Drive The amount to boost (or cut) the signal level to drive the distortion. Higher values will increase the distortion of high frequency signal components.

Dist Curve The curvature of the distortion. **0**% is no curvature (no distortion at all). At **100**%, the curve bends over smoothly and becomes perfectly flat right before it goes into digital clipping.

Highpass A first order highpass filter that removes low frequencies prior to being distorted.

975 HarmonicSuppress

Stereo algorithm to expand a single frequency band or harmonic bands.

PAUs: 2

HarmonicSuppress is a special expander algorithm. In most respects it is identical to **Expander**. However, **HarmonicSuppress** expands on only harmonically related bands of frequencies. Why would we do this? Imagine you are working with a sampled recording which contains an obnoxious 60 Hz hum. You can suppress just the 60 Hz component witha parametric filter. Set the threshold above the level of the 60 Hz tone so that real sounds in the 60 Hz range above threshold pass through the expander while 60 Hz hum below threshold get squashed. If the 60 Hz is more of a hum or buzz than a tone, then distortion harmonics are present. In this case, you might have more success using **HarmonicSuppress**.

HarmonicSuppress is based on comb filtering—a simple filter which removes harmonically related frequency bands with a spectrum which looks like a comb. With the Harmonics parameter, you can choose to expand the odd harmonics (including the fundamental) or even harmonics (not including the fundamental) or all harmonics. (Choosing all harmonics is the same as choosing even harmonics at half the frequency.

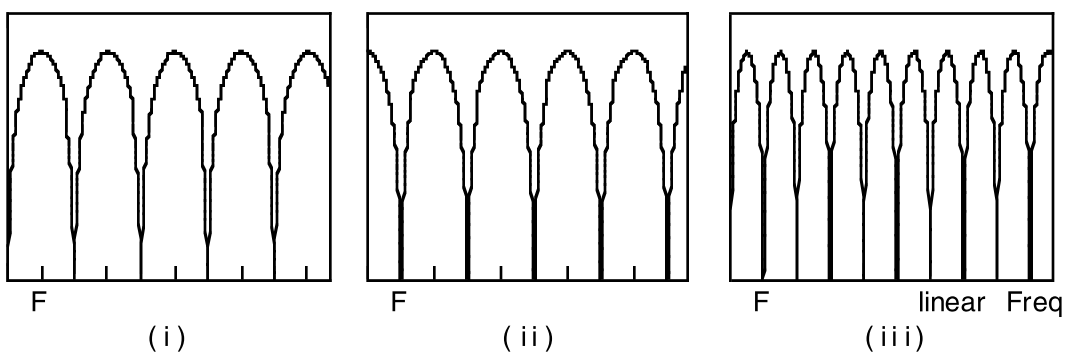


Figure 128 HarmonicSuppress filtering at full expansion
F marks fundamental
Harmonics are Even (i), Odd (ii), All (iii)

The algorithms expand the signal in the specified band(s) (reduce the signal's gain) when the signal falls below the expansion threshold in the specified band(s). You can select which channel, left (**L**), right (**R**) or the larger of the two (**L & R**) is used to control the expansion (side chain processing) with the SC Input parameter. You can also select which channel is actually expanded, again left (**L**), right (**R**) or both (**L & R**) using the ExpandChan parameter. 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, but that's a lot.) 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.

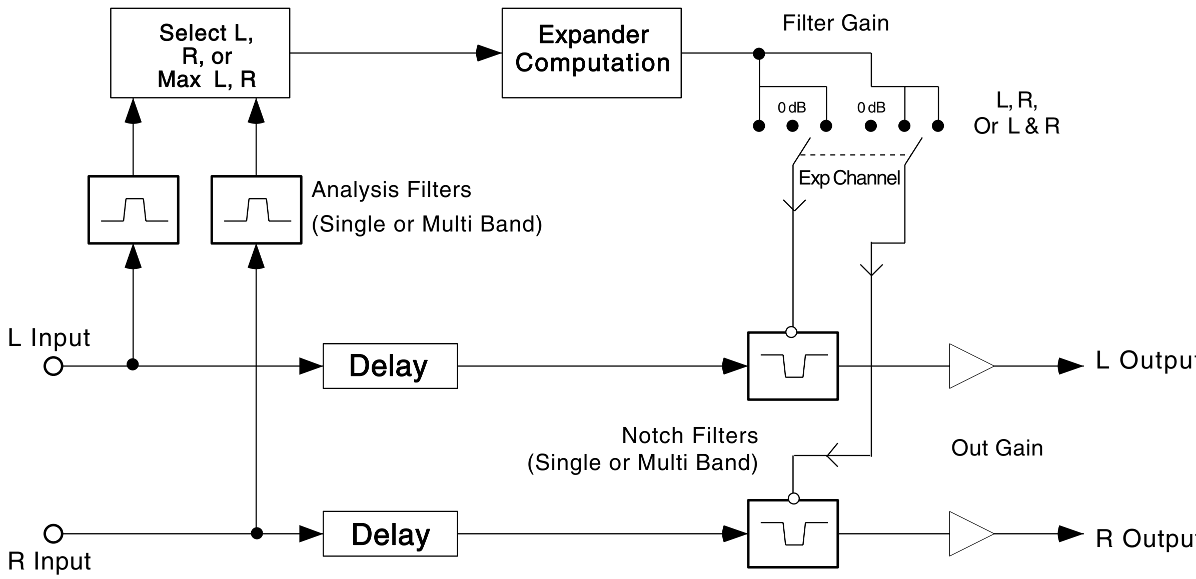


Figure 129 Band suppression

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.

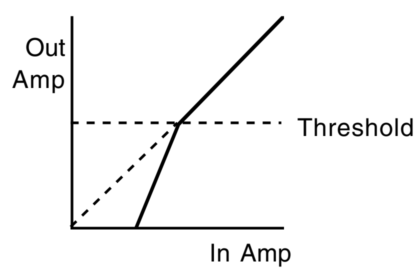


Figure 130 Expansion transfer characteristic

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 (HarmonicSuppress)

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Harmonics	Even, Odd, All		
Fund FreqC	8 to 25088 Hz	SC Input	L, R, L & R
Fund FreqF	-100 to 100 ct	ExpandChan	L, R, L & R

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									
					|||||||||||||||||||||||||||||				Reduction	
	-dB	40	20	12	8	6	4	2	0	

In/Out When set to **In** the expander is active; when set to **Out** the band suppressor 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**.

Fund FreqC The coarse frequency control sets the fundamental frequency of the harmonic structure to be expanded. Since the filter is a comb filter, the separation between harmonically related expansion bands is also controlled.

Fund FreqF The fine frequency control sets the fundamental frequency of the harmonic structure to be expanded. Since the filter is a comb filter, the separation between harmonically related expansion bands is also controlled.

Harmonics Sets the harmonic structure of the expansion comb filter. When set to **Even**, only the even harmonics of the specified fundamental frequency (including any dc signal level) are expanded, with no expansion of the fundamental. When set to **Odd**, the odd harmonics, including the fundamental, are expanded. The **All** setting expands all even and odd harmonics including any dc signal level. The **All** setting is the same as the **Even** when the **Even** frequency is set to half the value of **All**.

SC Input Select the input source channel for side-chain processing—left (**L**), right (**R**) or both (**L & R)**. When set to **L & R**, the maximum of left and right amplitudes is used.

ExpandChan Select which input channel will receive expander processing—left, right or both. If you select left or right, the opposite channel will pass through unaffected.

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 predelay). 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.

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 K2661, 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 K2661'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 K2661'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 K2661's control sources.

DSP Digital signal processing (see).

DSP Functions The K2661'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 K2661. 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 K2661'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 K2661 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 K2661 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 K2661, 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 K2661'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 (Mast2 page) to do a hard reset. This is a quick way to restore the factory defaults to your K2661, 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 K2661 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 K2661 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 K2661's 48 available voices. Each K2661 program can contain up to 32 layers. Any program can contain up to 32 layers. Ones with more than three show up in display as "Drum" programs.

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 K2661'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 K2661 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 K2661'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 K2661'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 K2661'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 K2661's basic performance-level sound object. Programs can consist of up to (32 layers; 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 K2661 they are stored in sample RAM (and are not saved across power cycles). When you save a program you've created, you're writing to program RAM (P/RAM). P/RAM is battery-backed, so it is saved from session to session. 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 K2661 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. A SCSI device—a computer, hard disk, printer, just about anything that sends or receives information in standardized form—is 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 K2661 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 K2661 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 K2661's operation while in Setup mode, as well as determining the Program Change numbers and controller messages the K2661 sends via MIDI.

Soft Reset Returns the K2661 to Program mode without affecting the contents of RAM. Press the **+/-**, **0**, and **Clear** buttons to do a soft reset. If your K2661 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 K2661, 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.

Chapter 12
Triple Modular Processing

Overview

Triple-modular processing is an enhancement to Kurzweil's VAST synthesis model (Variable-Architecture Synthesis Technology). The VAST model incorporates multi-layer programs; each layer in a VAST program uses an algorithm that includes pitch control, the renowned Kurzweil VAST DSP (digital signal-processing) functions, and amplitude control. This provides for a huge number of combinations.

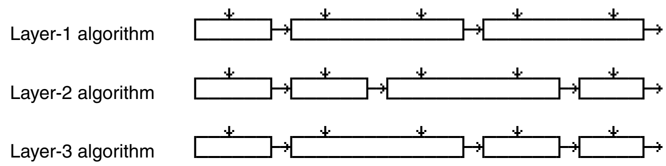


Figure 12-1 Algorithms of three possible normal layers, with three separate signal paths

Triple-modular processing makes the VAST model even more flexible, enabling you to create *triple layers* (or just *triples*)—specialized groups of three layers with one continuous signal path. The audio signal from the first layer of a triple gets routed to the second layer, where it's mixed with whatever audio signal the second layer may be generating. Similarly, the output of the second layer is routed to the third layer. The third layer's output goes to the input buses of KDFX, then to the physical audio outputs.

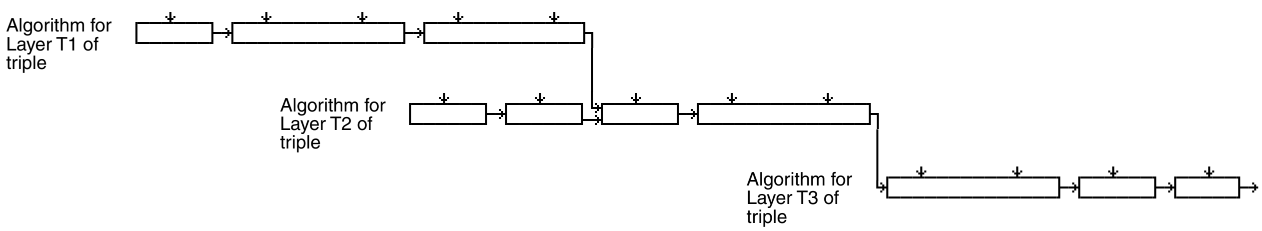


Figure 12-2 Algorithm of a possible triple, with a *single* signal path

This enables you to create sophisticated signal paths and chains of DSP functions. For example, you can mix the audio signal generated by one layer with the audio signal generated by the following layer, or you can apply three layers of DSP to the audio signal generated by Layer 1. There's an almost limitless number of possible combinations (something like 30 *billion*, depending on how you count).

In Figure 12-2, for example, the processed audio signal from Layer 1 (sample playback and/or one or more DSP-generated waveforms) gets mixed with the processed audio signal from Layer 2. The Layer-1 signal gets mixed with the Layer-2 signal just before the third DSP block. If there were a different algorithm in Layer 2, the Layer-1 signal might get mixed with the Layer-2 signal at a different location, or in multiple locations (both Layer 2 and Layer 3 can have multiple inputs from the previous layer).

Note that any processing in Layer 2 that's "downstream" (to the right) of the input(s) from Layer 1 gets applied to the signals from both Layer 1 and Layer 2. After the processing in Layer 2, the signal passes to Layer 3 for still more processing before going to KDFX.

Any non-KB3 program can contain a triple, alone or in combination with normal layers and/or other triples (KB3 programs don't use layers, so they can't contain triples). Each triple is a contiguous set of three otherwise normal layers, with all the usual layer parameters. Because their audio paths are linked, the layers of a triple are always numbered consecutively. You can't reorder the layers within a triple, but you can locate the triple anywhere within a program. For example, a ten-layer program might contain two triples: one triple spanning Layers 2–4 and the other spanning Layers 7–9 (with Layers 1, 5, 6, and 10 being normal layers).

Triples and Polyphony

Each triple uses three voices, the equivalent of three normal layers. By using a triple instead of three normal layers, you add millions of sound-editing possibilities without using any more voices. If you play enough notes to exceed the K2661's polyphonic limit, the three voices used by each triple get stolen as a unit, according to the priority set by the parameters on the AMPENV page of Layer 3 of the triple.

Soloing and Muting

You can solo and mute triples much as you can normal layers. Soloing and muting always affects the entire triple, however; you can't solo or mute individual layers of a triple.

KB3 Programs

You can't use triples in KB3 programs, since KB3 programs don't use the VAST model. That is, they don't consist of layers like non-KB3 programs (instead, every KB3 program plays notes that are generated by tone wheels that start running as soon as you select the program). Consequently, KB3 programs can't contain triples, because triples are specialized layers.

Even so, KB3 programs can coexist in setups with programs that use triples. Keep in mind though, that since a typical KB3 program uses about 40 voices, any setup that uses one of these programs has only eight voices left over. This means that you might not always hear every note of a setup that contains both a KB3 program and one or more programs containing a triple (or multiple triples). Whenever a KB3 note is playing, the K2661 can play a maximum of two notes from triples.

***Note**: You can edit a KB3 program to reduce the number of tone wheels it uses, which frees up additional voices (one voice for every two tone wheels).*

Live Mode

Live Mode enables you to take an external audio signal and route it through the K2661's VAST sound engine. (This works only if you have the Sampling Option installed in your K2661.) Triple-modular processing makes for an even greater number of possibilities in Live mode. For example, a triple can route a single Live-mode input through three layers of DSP, or it can mix a Live-mode input with an internally-generated K2661 sound. A triple can also mix two Live-mode inputs with each other and/or with an internally-generated K2661 sound.

Algorithms for Triple Modular Processing

There are 94 new VAST algorithms, incorporating familiar Kurzweil DSP functions. The algorithms are divided into three sets, each of which correspond to a specific layer within the triple. In other words, certain algorithms are available only for certain layers.

Layer-1 Algorithms: 33–62

There are 30 algorithms available for Layer 1 of any triple. The output of each of these algorithms is permanently configured to go to one or more inputs in a Layer-2 algorithm (63–100). All of these algorithms begin with pitch blocks, like the normal VAST algorithms. This ensures that every triple has an audio signal.

Layer-2 Algorithms: 63–100

There are 38 algorithms available for Layer 2 of any triple. The output of each of these algorithms is permanently configured to go to one or more inputs in a Layer-3 algorithm (101–127).

Some of the Layer-2 algorithms (63–80) begin with pitch blocks, and some (81–100) do not. Layers that use algorithms 81–100 don't make use of the parameters on the PITCH page in the Program Editor.

Layer-3 Algorithms: 101–126

There are 26 algorithms available for Layer 3 of any triple. The output of each of these algorithms is permanently configured to go to the input of KDFX. None of these algorithms has a pitch block, so layers that use them don't make use of the parameters on the PITCH page in the Program Editor.

Algorithm Reference

Turn to page 12-12 for diagrams of each algorithm, including the DSP functions available for each block in each algorithm.

Compatibility with Other Kurzweil Instruments

K2000s and K2500s (as well as all earlier models) can't load programs that contain triples, since they don't have the software required to support triples. Similarly, you can't load programs containing triples into K2600s with operating system versions earlier than v2.0.

Creating Triples

There are several ways to add a triple to a program. The most convenient method depends on what you want to do.

In case you're wondering, there's no way to convert a normal layer to a triple. Whenever you want to create a program containing a triple, or add a triple to a program, use one of the following methods.

Creating a New Triple

Creating a new triple inserts a copy of Layers 1–3 of Program **739** into the current program. This enables you to use Program **739** as a template for triples.

1. While in Program mode, select the program in which you want to create a triple.
2. Press **Edit** to enter the Program Editor.
3. Press **<more**, then press **NewLyr**. This displays a prompt asking you whether to create a normal layer or a triple.
4. Press **Triple**. The K2661 creates the triple, adding three linked layers after the highest-numbered existing layer (for example, if the program contained three layers, the new triple would occupy Layers 4–6). Adding the triple takes a few seconds; the display shows you the K2661's progress).
5. Rename and/or save the program as desired.

If you want to use one of your own triples as the template, edit the triple in Program **739**, then save the program with ID **739** (this creates a RAM version of the program, temporarily overwriting the ROM version—you can restore the ROM version by deleting Program **739**, which erases the RAM version).

Copying a Program Containing a Triple

This method works well when you want to make a few relatively simple adjustments to an existing program (without changing the original): changing keymaps, using a different KDFX studio, tweaking an amplitude envelope, or adjusting one or more DSP blocks in the triple itself.

1. While in Program mode, select the program you want to copy.
2. Press **Edit** to enter the Program Editor.
3. Press **<more** until you see the **Name** soft button.
4. Press **Name**, rename the program, and press **OK**.
5. Press **Save**, and save the program to a new ID (if you're editing a RAM program, change the ID to avoid replacing an existing program).
6. Edit the new program as desired.

As an alternative, you can use the Copy function in Master mode, as described in your Musician's Guide.

Importing a Triple into a Program

Use this method when there's an existing triple that you want to insert into a particular program.

1. While in Program mode, select the program into which you want to import the triple.
2. Press **Edit** to enter the Program Editor.
3. Press **<more**, then press **ImpLyr** (Import Layer).
4. When the Import Layer dialog appears, select a program using the **Left** and **Right** cursor buttons.
5. Select a triple using the **Up** and **Down** cursor buttons. As you scroll through the layers, the display indicates which layers are triples. Notice that you can't select individual layers within a triple.
6. Press **Import**. The K2661 imports the specified triple, adding three linked layers after the highest-numbered existing layer.
7. Rename and/or save the program as desired.

Editing Triples

The ALG page contains all the parameters for editing triples. The following diagram shows the ALG page for Layer 2 of Program **739 Default Triple**:

Notice the arrow pointing to the left side of the first DSP block (not the arrows pointing to the top of the block; these are the DSP block's control inputs). The arrow pointing to the left side of the DSP block represents the output from the first layer of the triple, which is the input for the second layer of the triple.

The third line of the display clearly identifies the layer as part of a triple. The top line also indicates that you're looking at a triple: T2 indicates that the current layer is the first layer of a triple, while 2/3 indicates that the current layer is Layer 2 of three layers in the program. Similarly, LyrT3:6/8 would indicate that the current layer is Layer 3 of a triple, and is the sixth layer in a program containing a total of eight layers.

There are four basic operations in editing a triple:

- Selecting a layer with the **Chan/Bank** buttons
- Selecting an algorithm
- Changing DSP functions within the DSP blocks of an algorithm
- Editing the control page for each DSP block in each layer

The parameters on the ALG page consist of the algorithm itself, and the DSP blocks within the algorithm (In the diagram, the second DSP block in the algorithm is selected). Use the cursor buttons to select the parameters, and a data entry method to change the value of the selected parameter.

Amplitude Envelopes

A layer's amplitude envelope is a critical element of the layer's sound; it determines the layer's attack, decay, and release times—among other things. These features are controlled in one of two ways: by default settings stored in a layer's keymap, or by the settings for the parameters on the layer's AMPENV page.

The amplitude envelope gets applied to a layer's audio signal as it passes through the algorithm's AMP block (if it contains one). All the normal VAST algorithms (1–31) end with an AMP block, so normal VAST layers always use an amplitude envelope. This isn't true for all triple algorithms, however.

All the Layer-3 algorithms (101–126) end with AMP blocks, which ensures that there's at least one amplitude envelope for every triple. For Layers 1 and 2 of a triple, the AMP block is optional. To include an AMP block in an algorithm, assign a value of **AMP** in any one-stage DSP block from F1 to F4 (the PITCH block, if the layer contains one, has a fixed value).

It's important to keep in mind that whatever happens in the amplitude envelops in Layers 1 and 2 of a triple, the amplitude envelope in layer 3 affects the sound of the entire triple. For example, if you have a long release in the amplitude envelope in Layer 1, but a short release in the amplitude envelope in Layer 3, Layer 1's release will continue only as long as Layer 3's release

AMP Blocks in Layers 1 and 2 of a Triple

When you put an AMP block in Layer 1 or 2 of a triple, that layer uses its own amplitude envelope, as specified on the layer's AMPENV page. This envelope affects only its own layer, and not the subsequent layers in the triple. Also, amplitude envelopes (if any) applied to Layers 1 and 2 don't apply MIDI volume data; only the amplitude envelope in Layer 3 controls overall volume.

A layer that doesn't contain an AMP block ignores its own amplitude envelope. So, for example, if Layer 1 of a triple has no AMP block, Layer 1's output has no amplitude envelope. As the signal from Layer 1 mixes with the signal from Layer 2, the amplitude envelope from Layer 2, if any, may affect the sound from Layer 1 (this depends on Layer 2's inputs—see *Input Locations* on page 12-7.

When to Use AMP Blocks in Layers 1 and 2

Put an AMP block in Layer 1 or Layer 2 whenever you want that layer to use its own amplitude envelope. This affects the layer's sound to varying degrees, depending on two things: the keymap assigned to the layer, and the mode (Natural or User) of the layer's AMPENV.

AMP blocks can be particularly important in Layers 1 and 2 when those layers use keymaps that contain ROM samples or edited versions of ROM samples (especially samples of acoustic sounds like piano or strings). That's because the factory-default (natural) amplitude envelopes of many of these keymaps also control the decompression of the keymap's samples (to maximize audio quality, many samples are stored in compressed form, and are decompressed just before playback). If there's no AMP block, there's no amplitude envelope, and therefore no decompression information. This can alter the sound of a ROM sample drastically.

When Layer 1 or 2 of a triple uses a keymap containing ROM samples (or edited ROM samples), and you want the samples to sound relatively normal, you'll generally need to do two things:

- Include an AMP block in the layer
- Use the layer's default (natural) amplitude envelope

The AMP block ensures that the layer's amplitude envelope gets applied to the layer. The natural amplitude envelope ensures that the decompression information gets applied to the layer (user-defined envelopes contain only Attack-Decay-Release information, and no decompression information). Of course, you might like the way a layer sounds *without* the AMP block, or with a user-defined amplitude envelope.

To check whether a layer's keymap uses its default (natural) amplitude envelope or a user-defined amplitude envelope, go to the AMPENV page for that layer. If the only visible parameter is Mode, with a value of **Natural**, the layer is using its natural amplitude envelope. If you see a lot of parameters like Att1, Dec1, and Rel1, the layer is using a user-defined amplitude envelope. To revert to the natural amplitude envelope, select the word **User**, and change it to **Natural** by turning the Alpha Wheel one click in either direction.

Input Locations

We mentioned that Layer 2's amplitude envelope may or may not affect the sound of Layer 1—it depends on the location of Layer 2's AMP block relative to the input(s) from Layer 1.

Figure 12-3 shows the algorithm for Layer 2 of a triple. The input point from Layer 1 is in the third DSP block (F2, a crossfade mixer in this case), as indicated by the arrow entering the F2 block from above the algorithm. Layer 2's AMP block (F1, which is highlighted) is *before* the input point, which means that the amplitude envelope for Layer 2 gets applied *before* the signal from Layer 1 joins the signal from Layer 2. Consequently the amplitude envelope of Layer 2 affects only Layer 2, and doesn't affect the sound of Layer 1.

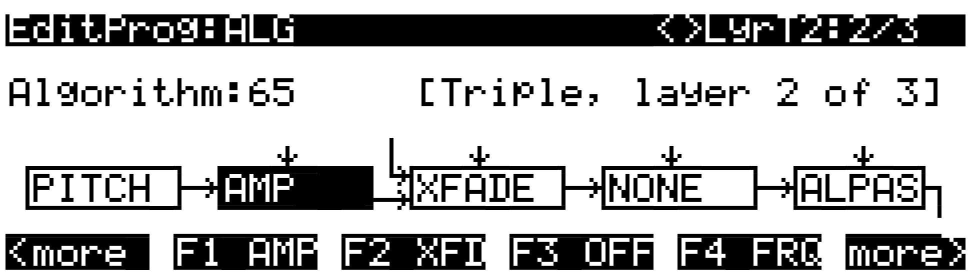


Figure 12-3 AMP block before input point

Figure 12-4 shows the same algorithm, with one change: the AMP block is now in F3 (highlighted), *after* the input point of Layer 2. This means that the amplitude envelope for

Layer 2 gets applied *after* the signal from Layer 1 joins the signal from Layer 2. Consequently, Layer 2's amplitude envelope affects Layer 1's sound as well as Layer 2's sound.

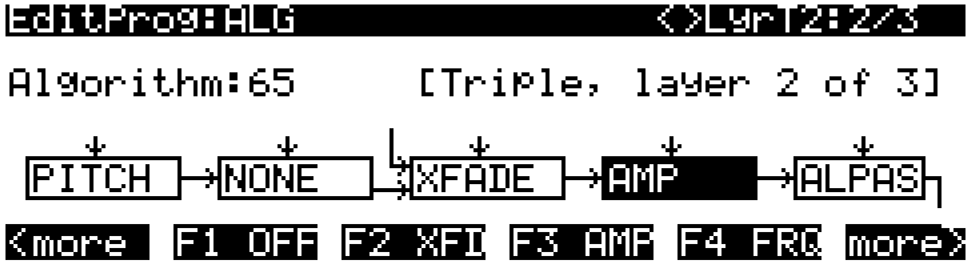


Figure 12-4 AMP block after input point

Amplitude Envelopes and the Keymap in Layer 3

Since Layer 3 of a triple always ends with an AMP block, triples always apply Layer 3's amplitude envelope, which may override the effects of any amplitude envelopes in Layers 1 or 2. The effect of Layer 3's envelope on the sound of the triple depends on whether Layer 3 uses a natural amplitude envelope or a user-defined one. When Layer 3 of a triple uses a natural amplitude envelope, the keymap used in Layer 3 determines the final amplitude envelope for the triple (because the natural amplitude envelope is stored as part of the keymap). In other words, changing the keymap for Layer 3 changes the amplitude envelope. This is an easy way to use alternative amplitude envelopes—for example, applying the amplitude envelope of a piano to a guitar sound generated in layer 1 or Layer 2. If you use a user envelope in Layer 3, it doesn't matter what the keymap is; changing the keymap doesn't change the amplitude envelope.

The Natural Amplitude Envelope

Look at the AMPENV page for Layer 3 of the triple. If you see the default (natural) amplitude envelope (that is, if you see only the Mode parameter with a value of **Natural**), the natural amplitude envelope of the keymap used in Layer 3 defines the final amplitude envelope for the triple. So if you assign a piano keymap to Layer 3, the piano keymap's amplitude envelope affects the sound of the entire triple—possibly overriding the amplitude envelopes of Layers 1 and 2. On the other hand, if you use a keymap that doesn't produce a sound (like **0 None** or **168 Silence**), the triple produces no sound at all.

***Note**: When you're using the Layer-3 keymap's default (natural) amplitude envelope, changing the value of the Xpose parameter on the KEYMAP page for Layer 3 changes the scaling of the amplitude envelope, and consequently, the sound of the triple. In general, raising the value of Xpose decreases the duration of each amplitude-envelope segment, while lowering the value of Xpose increases the duration of each segment.*

The User Envelope

You can use one of your own amplitude envelopes instead of the layer's natural amplitude envelope. On the AMPENV page for Layer 3, change the value of the Mode parameter from **Natural** to **User**. This removes the default amplitude envelope, and applies a user-defined envelope according to the values of the parameters on the AMPENV page (Att1, Dec1, Rel1, etc.). In this case, it doesn't matter which keymap you assign to Layer 3.

Summary of Amplitude Envelopes in Triples

Because of the special nature of triples and their interactions with amplitude envelopes, you'll need to give some thought to the amplitude.in each layer of the triples you program. There are three primary points to remember, each of which significantly affects the sound of a triple:

- The optional use of AMP blocks in Layers 1 and 2 (which determines whether Layers 1 and 2 use their own amplitude envelopes)
- The keymaps (if any) in each layer of a triple (which determine the sound produced by each layer)
- Choosing natural or user amplitude envelopes, especially in Layer 3

Other Considerations

Triples are a lot like normal layers, but there are a few important differences.

Processing-Only Layers

Layers that use Algorithms 81–127 can be processing-only layers—that is, they can provide additional processing for the audio output of other layers, without generating their own sounds. (They can provide additional processing *and* produce sound, as you'll see.)

Layer 1 of a triple usually generates sound via sample playback (unless it uses Keymap **0 None** or **168 Silence**). In other words, the sample playback of Layer 1 is routed to the input of Layer 1's algorithm. Consequently Layer 1 doesn't work as a processing-only layer. In addition to processing sample playback, most of the Layer-1 algorithms (33–62) can also generate waveforms within the algorithms themselves, using DSP functions like **SINE** and **SAW**.

For some of the Layer-2 algorithms (63–80) the input is the sample playback from Layer 2, mixed at one or more points with the output from Layer 1. For the remainder of the Layer-2 algorithms (81–100), the only input is the output from Layer 1; the sample playback from layer 2 (if any) is not routed to the inputs of these algorithms. Most of the Layer-2 algorithms can also generate waveforms within the algorithms themselves.

The input for all of the Layer-3 algorithms is the output from Layer 2. These algorithms never process the sample playback (if any) from Layer 3. Like the other algorithms, however, they can generate their own waveforms.

Since layers that use Algorithms 81–127 don't process their own sample playback data, they have no pitch parameters, and their algorithm diagrams contain no pitch blocks. When you're editing a layer that uses one of these algorithms, if you press the **PITCH** soft button, you'll see a blank page.

Layer Parameters

Since a triple is essentially a single layer (except that it uses three voices per note), it has a single set of layer parameters. You can view and edit these parameters by pressing the **LAYER** soft button when Layer 1 of the triple is current. If you press **LAYER** when Layer 2 or 3 is current, you see a blank page.

Output Parameters

Triples have a single set of output parameters controlling all three layers (again because triples are essentially single layers). You can view and edit the output parameters by pressing the **OUTPUT** soft button when Layer 3 of the triple is current. If you press **OUTPUT** when Layer 1 or 2 is current, you see a blank page.

New Combinations of DSP Functions

Seven new two-stage DSP functions combine a filter or double shaper with a gain function (the gain occurs *after* the filtering/shaping). One or more of these functions is available in most of the two-stage DSP blocks. They're listed below.

LOPAS2 GAIN	LP2RES GAIN
HIPAS2 GAIN	SHAPE2 GAIN
BAND2 GAIN	LPGATE GAIN
NOTCH2 GAIN	

These functions are equivalent to two single-stage blocks in a v1.0 algorithm—a block using a filter or shaper followed by a block using the GAIN function.

Stereo Keymaps

Because of sound-processing requirements, triples can't use stereo keymaps. Consequently, there's no Stereo parameter on the Keymap page when any layer of a triple is the current layer. Programs that use both triples and normal layers can use stereo keymaps for the normal layers.

Note Stealing in Triples

When you exceed the 48-voice polyphonic limit, the K2661 uses the AMPENV-page settings for Layer 3 of a triple to determine how the voices from that triple get stolen. Consequently, if you want to change how the triple's notes get stolen, you should edit the parameters on the AMPENV page for Layer 3 of the triple (as opposed to editing the AMPENV parameters for Layers 1 or 2). For example, decreasing the duration of the decay and/or release segments reduces your polyphony requirements.

Using DSP Waveforms

Keep in mind that the DSP waveforms (like SINE, SQR, SAW, and NOISE are typically 5 to 6 dB hotter than ROM samples. This can cause clipping in some of the filters used in triple algorithms. Consequently, you may want to reduce the input levels for filters that are processing DSP waveforms.

Using PWM

DSP blocks that use the PWM function are meant to be followed by blocks using the DIST function (this is true for normal VAST layers, as well). See your Musician's Guide for more information.

Using NOISE+

The DSP function called NOISE+ is available for many one-stage DSP blocks in Layer-2 and Layer-3 algorithms. This provides a convenient way to add noise at one or more points in a triple. When you use this function in a DSP block, it adds nearly-white noise—that is, an audio signal with nearly equal amplitude at all audible frequencies—to the existing signal (unless the existing signal has zero amplitude, in which case no noise gets added).

In order to function as designed, DSP blocks that use NOISE+ must be followed immediately by one of the DSP functions listed below. Otherwise, you won’t be able to attenuate the level of the added noise. If you use NOISE+ in the *last* DSP block of a Layer-2 algorithm, the *first* DSP block of Layer 3’s algorithm must be one of these functions.

NONE	+AMP
AMP	+GAIN
PANNER	XFADE
BAL AMP	

Exploring the Possibilities

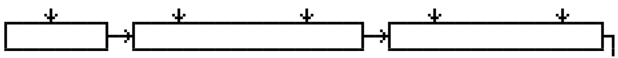
You may never exhaust the possible combinations of DSP functions available with triple-modular processing, but keep experimenting. We’ve found that using simple waveforms when trying different DSP functions makes it easier to hear the effects of each function. Try using SINE, SQR, and SAW in various DSP blocks. These block all incoming audio signals and generate a simple waveform as output (as opposed to SINE+, SQR+, and SAW+, which add waveforms to the existing signal).

Algorithm Reference

This section contains a diagram for each triple algorithm, as you see it in the K2661's display. Below each diagram is a list of the DSP functions available in each block of the algorithm.

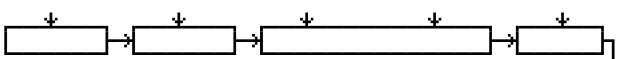
Layer-1 Algorithms (33–62)

Algorithm:33



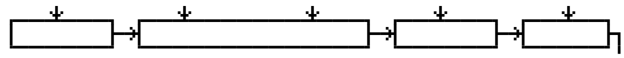
PITCH	NONE	NONE
	2PARAM SHAPER	2PARAM SHAPER
	LOPAS2 GAIN	LOPAS2 GAIN
	2POLE LOWPASS	2POLE LOWPASS
	BANDPASS FILT	BANDPASS FILT
	NOTCH FILTER	NOTCH FILTER
	2POLE ALLPASS	2POLE ALLPASS
	PARA BASS	PARA BASS
	PARA TREBLE	PARA TREBLE
	PARA MID	PARA MID

Algorithm:34



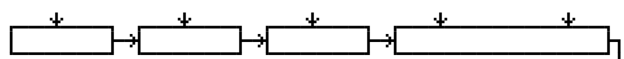
PITCH	NONE	NONE	NONE
	AMP	2PARAM SHAPER	AMP
	LOPASS	LOPAS2 GAIN	LOPASS
	HIPASS	2POLE LOWPASS	HIPASS
	ALPASS	BANDPASS FILT	ALPASS
	GAIN	NOTCH FILTER	GAIN
	SHAPER	2POLE ALLPASS	SHAPER
	DIST	PARA BASS	DIST
	PWM	PARA TREBLE	PWM
	SINE	PARA MID	SINE
	LF SIN		LF SIN
	SW+SHP		SW+SHP
	SAW+		SAW+
	SAW		SAW
	LF SAW		LF SAW
	SQUARE		SQUARE
	LF SQR		LF SQR
	WRAP		WRAP

Algorithm:35



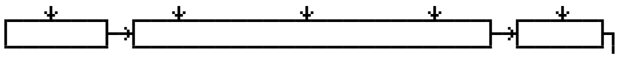
PITCH	NONE	NONE	NONE
	2PARAM SHAPER	AMP	AMP
	LOPAS2 GAIN	LOPASS	LOPASS
	2POLE LOWPASS	HIPASS	HIPASS
	BANDPASS FILT	ALPASS	ALPASS
	NOTCH FILTER	GAIN	GAIN
	2POLE ALLPASS	SHAPER	SHAPER
	PARA BASS	DIST	DIST
	PARA TREBLE	PWM	PWM
	PARA MID	SINE	SINE
		LF SIN	LF SIN
		SW+SHP	SW+SHP
		SAW+	SAW+
		SAW	SAW
		LF SAW	LF SAW
		SQUARE	SQUARE
		LF SQR	LF SQR
		WRAP	WRAP

Algorithm:36



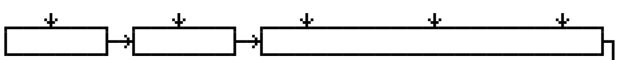
PITCH	NONE	NONE	NONE
	AMP	AMP	2PARAM SHAPER
	LOPASS	LOPASS	LOPAS2 GAIN
	HIPASS	HIPASS	2POLE LOWPASS
	ALPASS	ALPASS	BANDPASS FILT
	GAIN	GAIN	NOTCH FILTER
	SHAPER	SHAPER	2POLE ALLPASS
	DIST	DIST	PARA BASS
	PWM	PWM	PARA TREBLE
	SINE	SINE	PARA MID
	LF SIN	LF SIN	
	SW+SHP	SW+SHP	
	SAW+	SAW+	
	SAW	SAW	
	LF SAW	LF SAW	
	SQUARE	SQUARE	
	LF SQR	LF SQR	
	WRAP	WRAP	

Algorithm:37



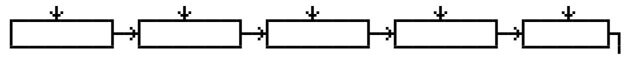
PITCH	NONE	NONE
	PARAMETRIC EQ	AMP
	STEEP RESONANT BASS	LOPASS
		HIPASS
		ALPASS
		GAIN
		SHAPER
		DIST
		PWM
		SINE
		LF SIN
		SW+SHP
		SAW+
		SAW
		LF SAW
		SQUARE
		LF SQR
		WRAP

Algorithm:38



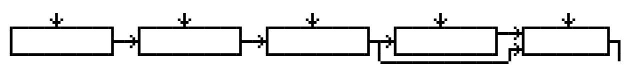
PITCH	NONE	NONE
	AMP	PARAMETRIC EQ
	LOPASS	STEEP RESONANT BASS
	HIPASS	
	ALPASS	
	GAIN	
	SHAPER	
	DIST	
	PWM	
	SINE	
	LF SIN	
	SW+SHP	
	SAW+	
	SAW	
	LF SAW	
	SQUARE	
	LF SQR	
	WRAP	

Algorithm:39



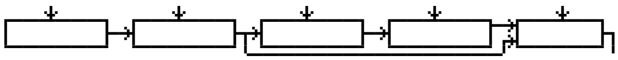
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	AMP	AMP
	LOPASS	LOPASS	LOPASS	LOPASS
	HIPASS	HIPASS	HIPASS	HIPASS
	ALPASS	ALPASS	ALPASS	ALPASS
	GAIN	GAIN	GAIN	GAIN
	SHAPER	SHAPER	SHAPER	SHAPER
	DIST	DIST	DIST	DIST
	PWM	PWM	PWM	PWM
	SINE	SINE	SINE	SINE
	LF SIN	LF SIN	LF SIN	LF SIN
	SW+SHP	SW+SHP	SW+SHP	SW+SHP
	SAW+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

Algorithm:40



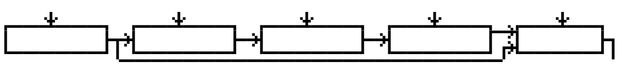
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	AMP	xGAIN
	LOPASS	LOPASS	LOPASS	+GAIN
	HIPASS	HIPASS	HIPASS	XFADE
	ALPASS	ALPASS	ALPASS	AMPMOD
	GAIN	GAIN	GAIN	
	SHAPER	SHAPER	SHAPER	
	DIST	DIST	DIST	
	PWM	PWM	PWM	
	SINE	SINE	SINE	
	LF SIN	LF SIN	LF SIN	
	SW+SHP	SW+SHP	SW+SHP	
	SAW+	SAW+	SAW+	
	SAW	SAW	SAW	
	LF SAW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

Algorithm:41



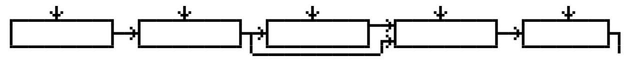
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	AMP	xGAIN
	LOPASS	LOPASS	LOPASS	+GAIN
	HIPASS	HIPASS	HIPASS	XFADE
	ALPASS	ALPASS	ALPASS	AMPMOD
	GAIN	GAIN	GAIN	
	SHAPER	SHAPER	SHAPER	
	DIST	DIST	DIST	
	PWM	PWM	PWM	
	SINE	SINE	SINE	
	LF SIN	LF SIN	LF SIN	
	SW+SHP	SW+SHP	SW+SHP	
	SAW+	SAW+	SAW+	
	SAW	SAW	SAW	
	LF SAW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

Algorithm:42



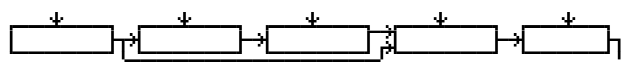
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	AMP	xGAIN
	LOPASS	LOPASS	LOPASS	+GAIN
	HIPASS	HIPASS	HIPASS	XFADE
	ALPASS	ALPASS	ALPASS	AMPMOD
	GAIN	GAIN	GAIN	
	SHAPER	SHAPER	SHAPER	
	DIST	DIST	DIST	
	PWM	PWM	PWM	
	SINE	SINE	SINE	
	LF SIN	LF SIN	LF SIN	
	SW+SHP	SW+SHP	SW+SHP	
	SAW+	SAW+	SAW+	
	SAW	SAW	SAW	
	LF SAW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

Algorithm:43



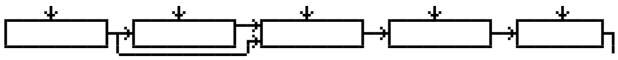
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	xGAIN	AMP
	LOPASS	LOPASS	+GAIN	LOPASS
	HIPASS	HIPASS	XFADE	HIPASS
	ALPASS	ALPASS	AMPMOD	ALPASS
	GAIN	GAIN		GAIN
	SHAPER	SHAPER		SHAPER
	DIST	DIST		DIST
	PWM	PWM		PWM
	SINE	SINE		SINE
	LF SIN	LF SIN		LF SIN
	SW+SHP	SW+SHP		SW+SHP
	SAW+	SAW+		SAW+
	SAW	SAW		SAW
	LF SAW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

Algorithm:44



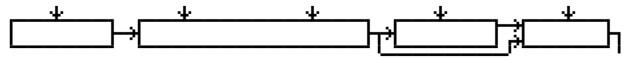
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	xGAIN	AMP
	LOPASS	LOPASS	+GAIN	LOPASS
	HIPASS	HIPASS	XFADE	HIPASS
	ALPASS	ALPASS	AMPMOD	ALPASS
	GAIN	GAIN		GAIN
	SHAPER	SHAPER		SHAPER
	DIST	DIST		DIST
	PWM	PWM		PWM
	SINE	SINE		SINE
	LF SIN	LF SIN		LF SIN
	SW+SHP	SW+SHP		SW+SHP
	SAW+	SAW+		SAW+
	SAW	SAW		SAW
	LF SAW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

Algorithm:45



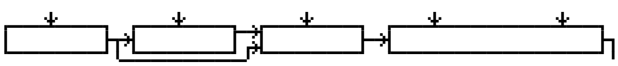
PITCH	NONE	NONE	NONE	NONE
	AMP	xGAIN	AMP	AMP
	LOPASS	+GAIN	LOPASS	LOPASS
	HIPASS	XFADE	HIPASS	HIPASS
	ALPASS	AMPMOD	ALPASS	ALPASS
	GAIN		GAIN	GAIN
	SHAPER		SHAPER	SHAPER
	DIST		DIST	DIST
	PWM		PWM	PWM
	SINE		SINE	SINE
	LF SIN		LF SIN	LF SIN
	SW+SHP		SW+SHP	SW+SHP
	SAW+		SAW+	SAW+
	SAW		SAW	SAW
	LF SAW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

Algorithm:47



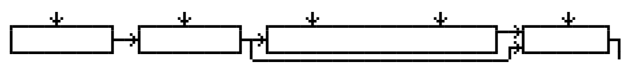
PITCH	NONE	NONE	NONE
	2PARAM SHAPER	AMP	xGAIN
	LOPAS2 GAIN	LOPASS	+GAIN
	2POLE LOWPASS	HIPASS	XFADE
	BANDPASS FILT	ALPASS	AMPMOD
	NOTCH FILTER	GAIN	
	2POLE ALLPASS	SHAPER	
	PARA BASS	DIST	
	PARA TREBLE	PWM	
	PARA MID	SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SAW	
		LF SAW	
		SQUARE	
		LF SQR	
		WRAP	

Algorithm:46



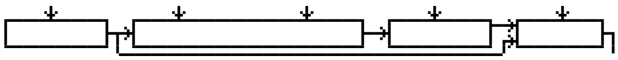
PITCH	NONE	NONE	NONE
	AMP	xGAIN	2PARAM SHAPER
	LOPASS	+GAIN	LOPAS2 GAIN
	HIPASS	XFADE	2POLE LOWPASS
	ALPASS	AMPMOD	BANDPASS FILT
	GAIN		NOTCH FILTER
	SHAPER		2POLE ALLPASS
	DIST		PARA BASS
	PWM		PARA TREBLE
	SINE		PARA MID
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		

Algorithm:48



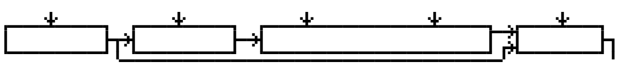
PITCH	NONE	NONE	NONE
	AMP	2PARAM SHAPER	xGAIN
	LOPASS	LOPAS2 GAIN	+GAIN
	HIPASS	2POLE LOWPASS	XFADE
	ALPASS	BANDPASS FILT	AMPMOD
	GAIN	NOTCH FILTER	
	SHAPER	2POLE ALLPASS	
	DIST	PARA BASS	
	PWM	PARA TREBLE	
	SINE	PARA MID	
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		

Algorithm:49



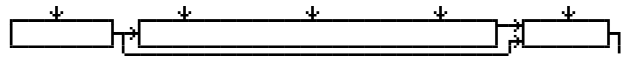
PITCH	NONE	NONE	NONE
	2PARAM SHAPER	AMP	xGAIN
	LOPAS2 GAIN	LOPASS	+GAIN
	2POLE LOWPASS	HIPASS	XFADE
	BANDPASS FILT	ALPASS	AMPMOD
	NOTCH FILTER	GAIN	
	2POLE ALLPASS	SHAPER	
	PARA BASS	DIST	
	PARA TREBLE	PWM	
	PARA MID	SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SAW	
		LF SAW	
		SQUARE	
		LF SQR	
		WRAP	

Algorithm:50



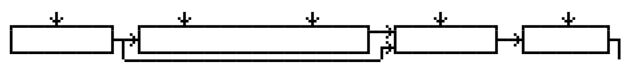
PITCH	NONE	NONE	NONE
	AMP	2PARAM SHAPER	xGAIN
	LOPASS	LOPAS2 GAIN	+GAIN
	HIPASS	2POLE LOWPASS	XFADE
	ALPASS	BANDPASS FILT	AMPMOD
	GAIN	NOTCH FILTER	
	SHAPER	2POLE ALLPASS	
	DIST	PARA BASS	
	PWM	PARA TREBLE	
	SINE	PARA MID	
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		

Algorithm:51



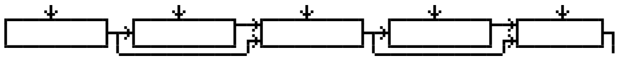
PITCH	NONE	NONE
	PARAMETRIC EQ	xGAIN
	STEEP RESONANT BASS	+GAIN
		XFADE
		AMPMOD

Algorithm:52



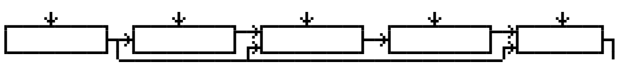
PITCH	NONE	NONE	NONE
	2PARAM SHAPER	xGAIN	AMP
	LOPAS2 GAIN	+GAIN	LOPASS
	2POLE LOWPASS	XFADE	HIPASS
	BANDPASS FILT	AMPMOD	ALPASS
	NOTCH FILTER		GAIN
	2POLE ALLPASS		SHAPER
	PARA BASS		DIST
	PARA TREBLE		PWM
	PARA MID		SINE
			LF SIN
			SW+SHP
			SAW+
			SAW
			LF SAW
			SQUARE
			LF SQR
			WRAP

Algorithm:53



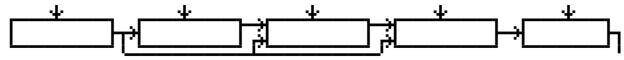
PITCH	NONE	NONE	NONE	NONE
	AMP	xGAIN	AMP	xGAIN
	LOPASS	+GAIN	LOPASS	+GAIN
	HIPASS	XFADE	HIPASS	XFADE
	ALPASS	AMPMOD	ALPASS	AMPMOD
	GAIN		GAIN	
	SHAPER		SHAPER	
	DIST		DIST	
	PWM		PWM	
	SINE		SINE	
	LF SIN		LF SIN	
	SW+SHP		SW+SHP	
	SAW+		SAW+	
	SAW		SAW	
	LF SAW		LF SAW	
	SQUARE		SQUARE	
	LF SQR		LF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

Algorithm:54



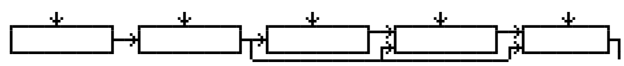
PITCH	NONE	NONE	NONE	NONE
	AMP	xGAIN	AMP	xGAIN
	LOPASS	+GAIN	LOPASS	+GAIN
	HIPASS	XFADE	HIPASS	XFADE
	ALPASS	AMPMOD	ALPASS	AMPMOD
	GAIN		GAIN	
	SHAPER		SHAPER	
	DIST		DIST	
	PWM		PWM	
	SINE		SINE	
	LF SIN		LF SIN	
	SW+SHP		SW+SHP	
	SAW+		SAW+	
	SAW		SAW	
	LF SAW		LF SAW	
	SQUARE		SQUARE	
	LF SQR		LF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

Algorithm:55



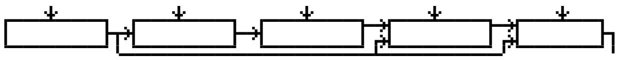
PITCH	NONE	NONE	NONE	NONE
	AMP	xGAIN	xGAIN	AMP
	LOPASS	+GAIN	+GAIN	LOPASS
	HIPASS	XFADE	XFADE	HIPASS
	ALPASS	AMPMOD	AMPMOD	ALPASS
	GAIN			GAIN
	SHAPER			SHAPER
	DIST			DIST
	PWM			PWM
	SINE			SINE
	LF SIN			LF SIN
	SW+SHP			SW+SHP
	SAW+			SAW+
	SAW			SAW
	LF SAW			LF SAW
	SQUARE			SQUARE
	LF SQR			LF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

Algorithm:56



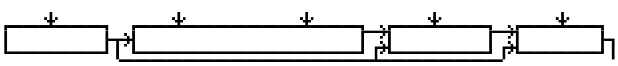
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	xGAIN	xGAIN
	LOPASS	LOPASS	+GAIN	+GAIN
	HIPASS	HIPASS	XFADE	XFADE
	ALPASS	ALPASS	AMPMOD	AMPMOD
	GAIN	GAIN		
	SHAPER	SHAPER		
	DIST	DIST		
	PWM	PWM		
	SINE	SINE		
	LF SIN	LF SIN		
	SW+SHP	SW+SHP		
	SAW+	SAW+		
	SAW	SAW		
	LF SAW	LF SAW		
	SQUARE	SQUARE		
	LF SQR	LF SQR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

Algorithm:57



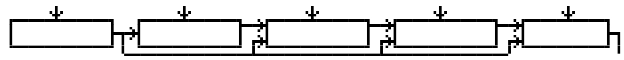
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	xGAIN	xGAIN
	LOPASS	LOPASS	+GAIN	+GAIN
	HIPASS	HIPASS	XFADE	XFADE
	ALPASS	ALPASS	AMPMOD	AMPMOD
	GAIN	GAIN		
	SHAPER	SHAPER		
	DIST	DIST		
	PWM	PWM		
	SINE	SINE		
	LF SIN	LF SIN		
	SW+SHP	SW+SHP		
	SAW+	SAW+		
	SAW	SAW		
	LF SAW	LF SAW		
	SQUARE	SQUARE		
	LF SQR	LF SQR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

Algorithm:58



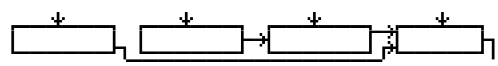
PITCH	NONE	NONE	NONE
	2PARAM SHAPER	xGAIN	xGAIN
	LOPAS2 GAIN	+GAIN	+GAIN
	2POLE LOWPASS	XFADE	XFADE
	BANDPASS FILT	AMPMOD	AMPMOD
	NOTCH FILTER		
	2POLE ALLPASS		
	PARA BASS		
	PARA TREBLE		
	PARA MID		

Algorithm:59



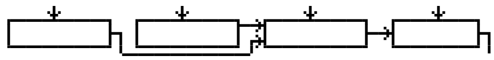
PITCH	NONE	NONE	NONE	NONE
	AMP	xGAIN	xGAIN	xGAIN
	LOPASS	+GAIN	+GAIN	+GAIN
	HIPASS	XFADE	XFADE	XFADE
	ALPASS	AMPMOD	AMPMOD	AMPMOD
	GAIN			
	SHAPER			
	DIST			
	PWM			
	SINE			
	LF SIN			
	SW+SHP			
	SAW+			
	SAW			
	LF SAW			
	SQUARE			
	LF SQR			
	WRAP			

Algorithm:60



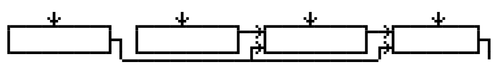
SYNCM	SYNCS	NONE	NONE
		AMP	xGAIN
		LOPASS	+GAIN
		HIPASS	XFADE
		ALPASS	AMPMOD
		GAIN	
		SHAPER	
		DIST	
		PWM	
		SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SAW	
		LF SAW	
		SQUARE	
		LF SQR	
		WRAP	

Algorithm:61



SYNCM	SYNCS	NONE	NONE
		xGAIN	AMP
		+GAIN	LOPASS
		XFADE	HIPASS
		AMPMOD	ALPASS
			GAIN
			SHAPER
			DIST
			PWM
			SINE
			LF SIN
			SW+SHP
			SAW+
			SAW
			LF SAW
			SQUARE
			LF SQR
			WRAP

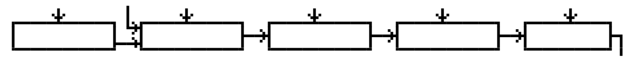
Algorithm:62



SYNCM	SYNCS	NONE	NONE
		xGAIN	xGAIN
		+GAIN	+GAIN
		XFADE	XFADE
		AMPMOD	AMPMOD

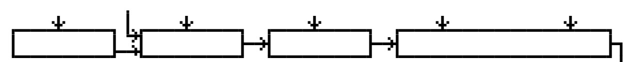
Layer-2 Algorithms (63–100)

Algorithm:63



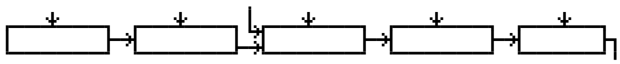
PITCH	NONE	NONE	NONE	NONE
	xGAIN	AMP	AMP	AMP
	+GAIN	LOPASS	LOPASS	LOPASS
	XFADE	HIPASS	HIPASS	HIPASS
	AMPMOD	ALPASS	ALPASS	ALPASS
		GAIN	GAIN	GAIN
		SHAPER	SHAPER	SHAPER
		DIST	DIST	DIST
		PWM	SINE	SINE
		SINE	LFSIN	LFSIN
		LF SIN	SW+SHP	SW+SHP
		SW+SHP	SAW+	SAW+
		SAW+	SW+DIST	SW+DIST
		SAW	LPCLIP	LPCLIP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

Algorithm:64



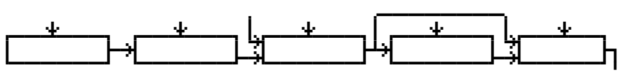
PITCH	NONE	NONE	NONE
	xGAIN	AMP	LOPAS2 GAIN
	+GAIN	LOPASS	HIPAS2 GAIN
	XFADE	HIPASS	BAND2 GAIN
	AMPMOD	ALPASS	NOTCH2 GAIN
		GAIN	LP2RES GAIN
		SHAPER	SHAPE2 GAIN
		DIST	LPGATE GAIN
		PWM	PARA MID
		SINE	
		LF SIN	
		SW+SHP	
		SAW+	
		SAW	
		LF SAW	
		SQUARE	
		LF SQR	
		WRAP	

Algorithm:65



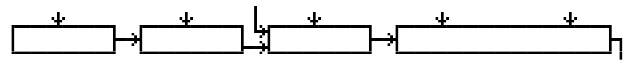
PITCH	NONE	NONE	NONE	NONE
	AMP	xGAIN	AMP	AMP
	LOPASS	+GAIN	LOPASS	LOPASS
	HIPASS	XFADE	HIPASS	HIPASS
	ALPASS	AMPMOD	ALPASS	ALPASS
	GAIN		GAIN	GAIN
	SHAPER		SHAPER	SHAPER
	DIST		DIST	DIST
	PWM		SINE	SINE
	SINE		LFSIN	LFSIN
	LF SIN		SW+SHP	SW+SHP
	SW+SHP		SAW+	SAW+
	SAW+		SW+DIST	SW+DIST
	SAW		LPCLIP	LPCLIP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

Algorithm:66



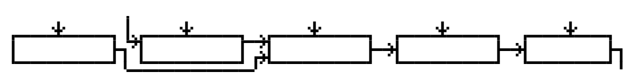
PITCH	NONE	NONE	NONE	NONE
	AMP	xGAIN	AMP	xGAIN
	LOPASS	+GAIN	LOPASS	+GAIN
	HIPASS	XFADE	HIPASS	!GAIN
	ALPASS	AMPMOD	ALPASS	
	GAIN		GAIN	
	SHAPER		SHAPER	
	DIST		DIST	
	PWM		SINE	
	SINE		LFSIN	
	LF SIN		SW+SHP	
	SW+SHP		SAW+	
	SAW+		SW+DIST	
	SAW		LPCLIP	
	LF SAW		SINE+	
	SQUARE		NOISE+	
	LF SQR			
	WRAP			

Algorithm:67



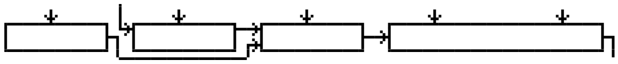
PITCH	NONE	NONE	NONE
	AMP	xGAIN	LOPAS2 GAIN
	LOPASS	+GAIN	HIPAS2 GAIN
	HIPASS	XFADE	BAND2 GAIN
	ALPASS	AMPMOD	NOTCH2 GAIN
	GAIN		LP2RES GAIN
	SHAPER		SHAPE2 GAIN
	DIST		LPGATE GAIN
	PWM		PARA MID
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		

Algorithm:68



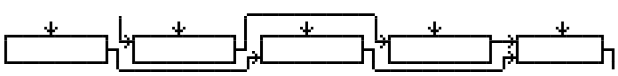
PITCH	NONE	NONE	NONE	NONE
	AMP	xGAIN	AMP	AMP
	LOPASS	+GAIN	LOPASS	LOPASS
	HIPASS	XFADE	HIPASS	HIPASS
	ALPASS	AMPMOD	ALPASS	ALPASS
	GAIN		GAIN	GAIN
	SHAPER		SHAPER	SHAPER
	DIST		DIST	DIST
	PWM		SINE	SINE
	SINE		LFSIN	LFSIN
	LF SIN		SW+SHP	SW+SHP
	SW+SHP		SAW+	SAW+
	SAW+		SW+DIST	SW+DIST
	SAW		LPCLIP	LPCLIP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

Algorithm:69



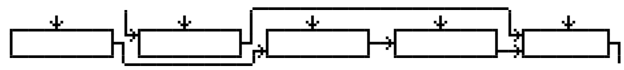
PITCH	NONE	NONE	NONE
	AMP	xGAIN	LOPAS2 GAIN
	LOPASS	+GAIN	HIPAS2 GAIN
	HIPASS	XFADE	BAND2 GAIN
	ALPASS	AMPMOD	NOTCH2 GAIN
	GAIN		LP2RES GAIN
	SHAPER		SHAPE2 GAIN
	DIST		LPGATE GAIN
	PWM		PARA MID
	SINE		
	LF SIN		
	SW+SHP		
	SAW+		
	SAW		
	LF SAW		
	SQUARE		
	LF SQR		
	WRAP		

Algorithm:70



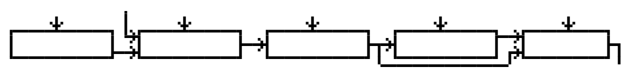
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	AMP	xGAIN
	LOPASS	LOPASS	LOPASS	+GAIN
	HIPASS	HIPASS	HIPASS	!GAIN
	ALPASS	ALPASS	ALPASS	
	GAIN	GAIN	GAIN	
	SHAPER	SHAPER	SHAPER	
	DIST	DIST	DIST	
	PWM	PWM	SINE	
	SINE	SINE	LFSIN	
	LF SIN	LF SIN	SW+SHP	
	SW+SHP	SW+SHP	SAW+	
	SAW+	SAW+	SW+DIST	
	SAW	SAW	LPCLIP	
	LF SAW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

Algorithm:71



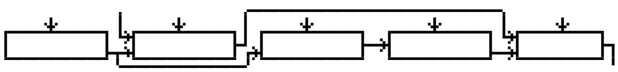
PITCH	NONE	NONE	NONE	NONE
	AMP	AMP	AMP	xGAIN
	LOPASS	LOPASS	LOPASS	+GAIN
	HIPASS	HIPASS	HIPASS	!GAIN
	ALPASS	ALPASS	ALPASS	
	GAIN	GAIN	GAIN	
	SHAPER	SHAPER	SHAPER	
	DIST	DIST	DIST	
	PWM	PWM	SINE	
	SINE	SINE	LFSIN	
	LF SIN	LF SIN	SW+SHP	
	SW+SHP	SW+SHP	SAW+	
	SAW+	SAW+	SW+DIST	
	SAW	SAW	LPCLIP	
	LF SAW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

Algorithm:72



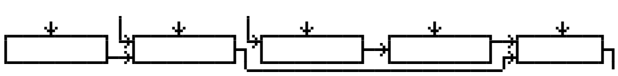
PITCH	NONE	NONE	NONE	NONE
	xGAIN	AMP	AMP	xGAIN
	+GAIN	LOPASS	LOPASS	+GAIN
	XFADE	HIPASS	HIPASS	!GAIN
	AMPMOD	ALPASS	ALPASS	
		GAIN	GAIN	
		SHAPER	SHAPER	
		DIST	DIST	
		PWM	SINE	
		SINE	LFSIN	
		LF SIN	SW+SHP	
		SW+SHP	SAW+	
		SAW+	SW+DIST	
		SAW	LPCLIP	
		LF SAW	SINE+	
		SQUARE	NOISE+	
		LF SQR		
	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

Algorithm:73



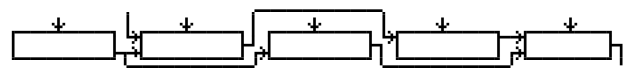
PITCH	NONE	NONE	NONE	NONE
	xGAIN	AMP	AMP	xGAIN
	+GAIN	LOPASS	LOPASS	+GAIN
	XFADE	HIPASS	HIPASS	!GAIN
	AMPMOD	ALPASS	ALPASS	
		GAIN	GAIN	
		SHAPER	SHAPER	
		DIST	DIST	
		PWM	SINE	
		SINE	LFSIN	
		LF SIN	SW+SHP	
		SW+SHP	SAW+	
		SAW+	SW+DIST	
		SAW	LPCLIP	
		LF SAW	SINE+	
		SQUARE	NOISE+	
		LF SQR		
	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

Algorithm:74



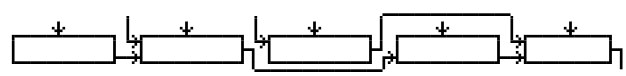
PITCH	NONE	NONE	NONE	NONE
	xGAIN	AMP	AMP	xGAIN
	+GAIN	LOPASS	LOPASS	+GAIN
	XFADE	HIPASS	HIPASS	!GAIN
	AMPMOD	ALPASS	ALPASS	
		GAIN	GAIN	
		SHAPER	SHAPER	
		DIST	DIST	
		PWM	SINE	
		SINE	LFSIN	
		LF SIN	SW+SHP	
		SW+SHP	SAW+	
		SAW+	SW+DIST	
		SAW	LPCLIP	
		LF SAW	SINE+	
		SQUARE	NOISE+	
		LF SQR		
	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

Algorithm:75



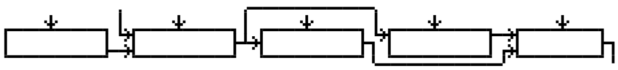
PITCH	NONE	NONE	NONE	NONE
	xGAIN	AMP	AMP	xGAIN
	+GAIN	LOPASS	LOPASS	+GAIN
	XFADE	HIPASS	HIPASS	!GAIN
	AMPMOD	ALPASS	ALPASS	
		GAIN	GAIN	
		SHAPER	SHAPER	
		DIST	DIST	
		PWM	SINE	
		SINE	LFSIN	
		LF SIN	SW+SHP	
		SW+SHP	SAW+	
		SAW+	SW+DIST	
		SAW	LPCLIP	
		LF SAW	SINE+	
		SQUARE	NOISE+	
		LF SQR		
	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

Algorithm:76



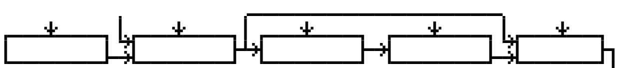
PITCH	NONE	NONE	NONE	NONE
	xGAIN	AMP	AMP	xGAIN
	+GAIN	LOPASS	LOPASS	+GAIN
	XFADE	HIPASS	HIPASS	!GAIN
	AMPMOD	ALPASS	ALPASS	
		GAIN	GAIN	
		SHAPER	SHAPER	
		DIST	DIST	
		PWM	SINE	
		SINE	LFSIN	
		LF SIN	SW+SHP	
		SW+SHP	SAW+	
		SAW+	SW+DIST	
		SAW	LPCLIP	
		LF SAW	SINE+	
		SQUARE	NOISE+	
		LF SQR		
	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

Algorithm:77



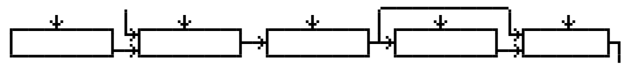
PITCH	NONE	NONE	NONE	NONE
	xGAIN	AMP	AMP	xGAIN
	+GAIN	LOPASS	LOPASS	+GAIN
	XFADE	HIPASS	HIPASS	!GAIN
	AMPMOD	ALPASS	ALPASS	
		GAIN	GAIN	
		SHAPER	SHAPER	
		DIST	DIST	
		PWM	SINE	
		SINE	LFSIN	
		LF SIN	SW+SHP	
		SW+SHP	SAW+	
		SAW+	SW+DIST	
		SAW	LPCLIP	
		LF SAW	SINE+	
		SQUARE	NOISE+	
		LF SQR		
	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

Algorithm:78



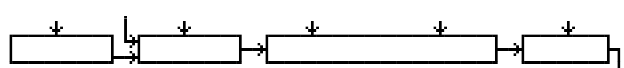
PITCH	NONE	NONE	NONE	NONE
	xGAIN	AMP	AMP	xGAIN
	+GAIN	LOPASS	LOPASS	+GAIN
	XFADE	HIPASS	HIPASS	!GAIN
	AMPMOD	ALPASS	ALPASS	
		GAIN	GAIN	
		SHAPER	SHAPER	
		DIST	DIST	
		PWM	SINE	
		SINE	LFSIN	
		LF SIN	SW+SHP	
		SW+SHP	SAW+	
		SAW+	SW+DIST	
		SAW	LPCLIP	
		LF SAW	SINE+	
		SQUARE	NOISE+	
		LF SQR		
	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

Algorithm:79



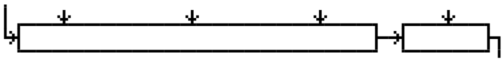
PITCH	NONE	NONE	NONE	NONE
	xGAIN	AMP	AMP	xGAIN
	+GAIN	LOPASS	LOPASS	+GAIN
	XFADE	HIPASS	HIPASS	!GAIN
	AMPMOD	ALPASS	ALPASS	
		GAIN	GAIN	
		SHAPER	SHAPER	
		DIST	DIST	
		PWM	SINE	
		SINE	LFSIN	
		LF SIN	SW+SHP	
		SW+SHP	SAW+	
		SAW+	SW+DIST	
		SAW	LPCLIP	
		LF SAW	SINE+	
		SQUARE	NOISE+	
		LF SQR		
	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

Algorithm:80



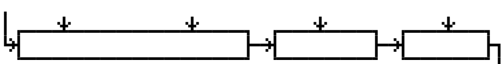
PITCH	NONE	NONE	GAIN
	xGAIN	PARA BASS	
	+GAIN	PARA TREBLE	
	XFADE		
	AMPMOD		

Algorithm:81



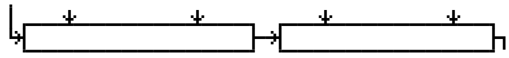
NONE	GAIN
HIFREQ STIMULATOR	
PARAMETRIC EQ	
STEEP RESONANT BASS	
4POLE LOPASS W/SEP	
4POLE HIPASS W/SEP	
TWIN PEAKS BANDPASS	
DOUBLE NOTCH W/SEP	

Algorithm:82



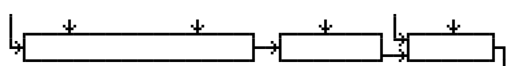
NONE	NONE	NONE
2PARAM SHAPER	AMP	AMP
LOPAS2 GAIN	LOPASS	LOPASS
2POLE LOWPASS	HIPASS	HIPASS
BANDPASS FILT	ALPASS	ALPASS
NOTCH FILTER	GAIN	GAIN
2POLE ALLPASS	SHAPER	SHAPER
PARA BASS	DIST	DIST
PARA TREBLE	SINE	SINE
PARA MID	LFSIN	LFSIN
	SW+SHP	SW+SHP
	SAW+	SAW+
	SW+DIST	SW+DIST
	LPCLIP	LPCLIP
	SINE+	SINE+
	NOISE+	NOISE+

Algorithm:83



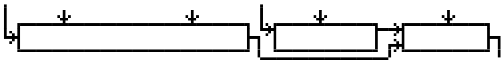
NONE	NONE
2PARAM SHAPER	LOPAS2 GAIN
LOPAS2 GAIN	HIPAS2 GAIN
2POLE LOWPASS	BAND2 GAIN
BANDPASS FILT	NOTCH2 GAIN
NOTCH FILTER	LP2RES GAIN
2POLE ALLPASS	SHAPE2 GAIN
PARA BASS	LPGATE GAIN
PARA TREBLE	PARA MID
PARA MID	

Algorithm:84



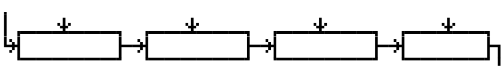
NONE	NONE	NONE
2PARAM SHAPER	AMP	xGAIN
LOPAS2 GAIN	LOPASS	+GAIN
2POLE LOWPASS	HIPASS	!GAIN
BANDPASS FILT	ALPASS	
NOTCH FILTER	GAIN	
2POLE ALLPASS	SHAPER	
PARA BASS	DIST	
PARA TREBLE	SINE	
PARA MID	LFSIN	
	SW+SHP	
	SAW+	
	SW+DIST	
	LPCLIP	
	SINE+	
	NOISE+	

Algorithm:85



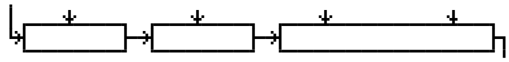
NONE	NONE	NONE
2PARAM SHAPER	AMP	xGAIN
LOPAS2 GAIN	LOPASS	+GAIN
2POLE LOWPASS	HIPASS	!GAIN
BANDPASS FILT	ALPASS	
NOTCH FILTER	GAIN	
2POLE ALLPASS	SHAPER	
PARA BASS	DIST	
PARA TREBLE	SINE	
PARA MID	LFSIN	
	SW+SHP	
	SAW+	
	SW+DIST	
	LPCLIP	
	SINE+	
	NOISE+	

Algorithm:86



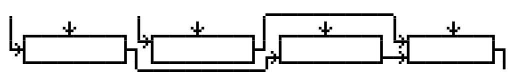
NONE	NONE	NONE	NONE
AMP	AMP	AMP	AMP
LOPASS	LOPASS	LOPASS	LOPASS
HIPASS	HIPASS	HIPASS	HIPASS
ALPASS	ALPASS	ALPASS	ALPASS
GAIN	GAIN	GAIN	GAIN
SHAPER	SHAPER	SHAPER	SHAPER
DIST	DIST	DIST	DIST
PWM	PWM	SINE	SINE
SINE	SINE	LFSIN	LFSIN
LF SIN	LF SIN	SW+SHP	SW+SHP
SW+SHP	SW+SHP	SAW+	SAW+
SAW+	SAW+	SW+DIST	SW+DIST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

Algorithm:87



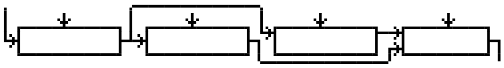
NONE	NONE	NONE
AMP	AMP	LOPAS2 GAIN
LOPASS	LOPASS	HIPAS2 GAIN
HIPASS	HIPASS	BAND2 GAIN
ALPASS	ALPASS	NOTCH2 GAIN
GAIN	GAIN	LP2RES GAIN
SHAPER	SHAPER	SHAPE2 GAIN
DIST	DIST	LPGATE GAIN
PWM	PWM	PARA MID
SINE	SINE	
LF SIN	LF SIN	
SW+SHP	SW+SHP	
SAW+	SAW+	
SAW	SAW	
LF SAW	LF SAW	
SQUARE	SQUARE	
LF SQR	LF SQR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

Algorithm:88



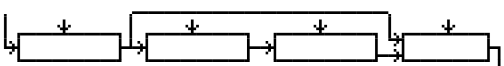
NONE	NONE	NONE	NONE
AMP	AMP	AMP	xGAIN
LOPASS	LOPASS	LOPASS	+GAIN
HIPASS	HIPASS	HIPASS	!GAIN
ALPASS	ALPASS	ALPASS	
GAIN	GAIN	GAIN	
SHAPER	SHAPER	SHAPER	
DIST	DIST	DIST	
PWM	PWM	SINE	
SINE	SINE	LFSIN	
LF SIN	LF SIN	SW+SHP	
SW+SHP	SW+SHP	SAW+	
SAW+	SAW+	SW+DIST	
SAW	SAW	LPCLIP	
LF SAW	LF SAW	SINE+	
SQUARE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

Algorithm:89



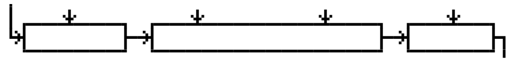
NONE	NONE	NONE	NONE
AMP	AMP	AMP	xGAIN
LOPASS	LOPASS	LOPASS	+GAIN
HIPASS	HIPASS	HIPASS	!GAIN
ALPASS	ALPASS	ALPASS	
GAIN	GAIN	GAIN	
SHAPER	SHAPER	SHAPER	
DIST	DIST	DIST	
PWM	PWM	SINE	
SINE	SINE	LFSIN	
LF SIN	LF SIN	SW+SHP	
SW+SHP	SW+SHP	SAW+	
SAW+	SAW+	SW+DIST	
SAW	SAW	LPCLIP	
LF SAW	LF SAW	SINE+	
SQUARE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

Algorithm:90



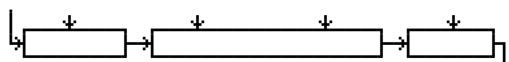
NONE	NONE	NONE	NONE
AMP	AMP	AMP	xGAIN
LOPASS	LOPASS	LOPASS	+GAIN
HIPASS	HIPASS	HIPASS	!GAIN
ALPASS	ALPASS	ALPASS	
GAIN	GAIN	GAIN	
SHAPER	SHAPER	SHAPER	
DIST	DIST	DIST	
PWM	PWM	SINE	
SINE	SINE	LFSIN	
LF SIN	LF SIN	SW+SHP	
SW+SHP	SW+SHP	SAW+	
SAW+	SAW+	SW+DIST	
SAW	SAW	LPCLIP	
LF SAW	LF SAW	SINE+	
SQUARE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

Algorithm:91



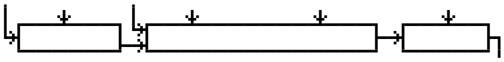
NONE	NONE	GAIN
AMP	PARA BASS	
LOPASS	PARA TREBLE	
HIPASS		
ALPASS		
GAIN		
SHAPER		
DIST		
PWM		
SINE		
LF SIN		
SW+SHP		
SAW+		
SAW		
LF SAW		
SQUARE		
LF SQR		
WRAP		

Algorithm:92



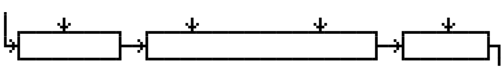
NONE	NONE	GAIN
AMP	SHAPE MOD OSC	
LOPASS	AMP MOD OSC	
HIPASS		
ALPASS		
GAIN		
SHAPER		
DIST		
PWM		
SINE		
LF SIN		
SW+SHP		
SAW+		
SAW		
LF SAW		
SQUARE		
LF SQR		
WRAP		

Algorithm:93



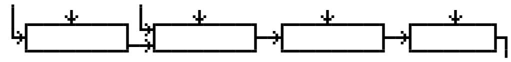
NONE	NONE	GAIN
AMP	x SHAPEMOD OSC	
LOPASS	+ SHAPEMOD OSC	
HIPASS		
ALPASS		
GAIN		
SHAPER		
DIST		
PWM		
SINE		
LF SIN		
SW+SHP		
SAW+		
SAW		
LF SAW		
SQUARE		
LF SQR		
WRAP		

Algorithm:94



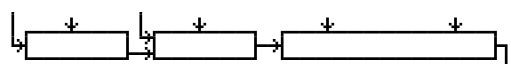
LOPAS2	SHAPEMOD OSC	GAIN

Algorithm:95



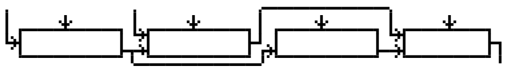
NONE	NONE	NONE	NONE
AMP	xGAIN	AMP	AMP
LOPASS	+GAIN	LOPASS	LOPASS
HIPASS	XFADE	HIPASS	HIPASS
ALPASS	AMPMOD	ALPASS	ALPASS
GAIN		GAIN	GAIN
SHAPER		SHAPER	SHAPER
DIST		DIST	DIST
PWM		SINE	SINE
SINE		LFSIN	LFSIN
LF SIN		SW+SHP	SW+SHP
SW+SHP		SAW+	SAW+
SAW+		SW+DIST	SW+DIST
SAW		LPCLIP	LPCLIP
LF SAW		SINE+	SINE+
SQUARE		NOISE+	NOISE+
LF SQR			
WRAP			

Algorithm:96



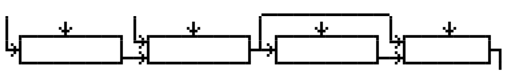
NONE	NONE	NONE
AMP	xGAIN	LOPAS2 GAIN
LOPASS	+GAIN	HIPAS2 GAIN
HIPASS	XFADE	BAND2 GAIN
ALPASS	AMPMOD	NOTCH2 GAIN
GAIN		LP2RES GAIN
SHAPER		SHAPE2 GAIN
DIST		LPGATE GAIN
PWM		PARA MID
SINE		
LF SIN		
SW+SHP		
SAW+		
SAW		
LF SAW		
SQUARE		
LF SQR		
WRAP		

Algorithm:97



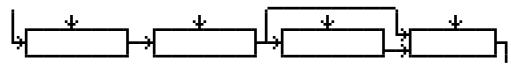
NONE	NONE	NONE	NONE
AMP	xGAIN	AMP	xGAIN
LOPASS	+GAIN	LOPASS	+GAIN
HIPASS	XFADE	HIPASS	!GAIN
ALPASS	AMPMOD	ALPASS	
GAIN		GAIN	
SHAPER		SHAPER	
DIST		DIST	
PWM		SINE	
SINE		LFSIN	
LF SIN		SW+SHP	
SW+SHP		SAW+	
SAW+		SW+DIST	
SAW		LPCLIP	
LF SAW		SINE+	
SQUARE		NOISE+	
LF SQR			
WRAP			

Algorithm:98



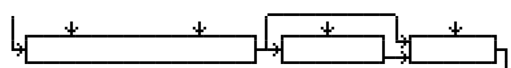
NONE	NONE	NONE	NONE
AMP	xGAIN	AMP	xGAIN
LOPASS	+GAIN	LOPASS	+GAIN
HIPASS	XFADE	HIPASS	!GAIN
ALPASS	AMPMOD	ALPASS	
GAIN		GAIN	
SHAPER		SHAPER	
DIST		DIST	
PWM		SINE	
SINE		LFSIN	
LF SIN		SW+SHP	
SW+SHP		SAW+	
SAW+		SW+DIST	
SAW		LPCLIP	
LF SAW		SINE+	
SQUARE		NOISE+	
LF SQR			
WRAP			

Algorithm:99



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

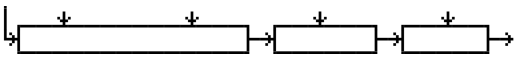
Algorithm:100



NONE	NONE	NONE
2PARAM SHAPER	AMP	xGAIN
LOPAS2 GAIN	LOPASS	+GAIN
2POLE LOWPASS	HIPASS	!GAIN
BANDPASS FILT	ALPASS	
NOTCH FILTER	GAIN	
2POLE ALLPASS	SHAPER	
PARA BASS	DIST	
PARA TREBLE	SINE	
PARA MID	LFSIN	
	SW+SHP	
	SAW+	
	SW+DIST	
	LPCLIP	
	SINE+	
	NOISE+	

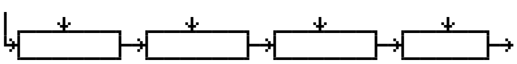
Layer-3 Algorithms (101–126)

Algorithm:101



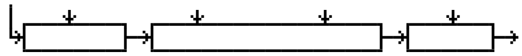
NONE	NONE	AMP
LOPAS2 GAIN	AMP	
HIPAS2 GAIN	LOPASS	
BAND2 GAIN	HIPASS	
NOTCH2 GAIN	ALPASS	
LP2RES GAIN	GAIN	
SHAPE2 GAIN	SHAPER	
LPGATE GAIN	DIST	
PARA MID	SINE	
	LFSIN	
	SW+SHP	
	SAW+	
	SW+DIST	
	LPCLIP	
	SINE+	
	NOISE+	

Algorithm:102



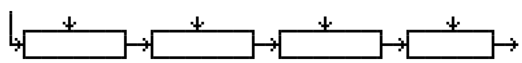
NONE	NONE	NONE	AMP
AMP	AMP	AMP	
LOPASS	LOPASS	LOPASS	
HIPASS	HIPASS	HIPASS	
ALPASS	ALPASS	ALPASS	
GAIN	GAIN	GAIN	
SHAPER	SHAPER	SHAPER	
DIST	DIST	DIST	
SINE	SINE	SINE	
LFSIN	LFSIN	LFSIN	
SW+SHP	SW+SHP	SW+SHP	
SAW+	SAW+	SAW+	
SW+DIST	SW+DIST	SW+DIST	
LPCLIP	LPCLIP	LPCLIP	
SINE+	SINE+	SINE+	
NOISE+	NOISE+	NOISE+	

Algorithm:103



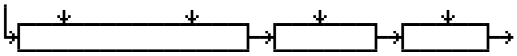
NONE	NONE	AMP
AMP	LOPAS2 GAIN	
LOPASS	HIPAS2 GAIN	
HIPASS	BAND2 GAIN	
ALPASS	NOTCH2 GAIN	
GAIN	LP2RES GAIN	
SHAPER	SHAPE2 GAIN	
DIST	LPGATE GAIN	
SINE	PARA MID	
LFSIN		
SW+SHP		
SAW+		
SW+DIST		
LPCLIP		
SINE+		
NOISE+		

Algorithm:104



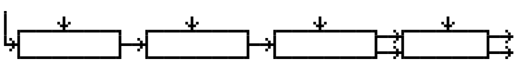
NONE	NONE	NONE	AMP
AMP	AMP	LOPAS2	
LOPASS	LOPASS	HIPAS2	
HIPASS	HIPASS	LPGATE	
ALPASS	ALPASS	LP2RES	
GAIN	GAIN	SHAPE2	
SHAPER	SHAPER		
DIST	DIST		
SINE	SINE		
LFSIN	LFSIN		
SW+SHP	SW+SHP		
SAW+	SAW+		
SW+DIST	SW+DIST		
LPCLIP	LPCLIP		
SINE+	SINE+		
NOISE+	NOISE+		

Algorithm:105



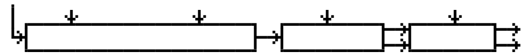
NONE	NONE	AMP
LOPAS2 GAIN	LOPAS2	
HIPAS2 GAIN	HIPAS2	
BAND2 GAIN	LPGATE	
NOTCH2 GAIN	LP2RES	
LP2RES GAIN	SHAPE2	
SHAPE2 GAIN		
LPGATE GAIN		
PARA MID		

Algorithm:106



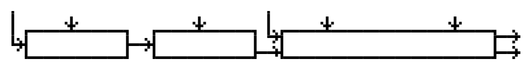
NONE	NONE	PANNER	AMP
AMP	AMP		
LOPASS	LOPASS		
HIPASS	HIPASS		
ALPASS	ALPASS		
GAIN	GAIN		
SHAPER	SHAPER		
DIST	DIST		
SINE	SINE		
LFSIN	LFSIN		
SW+SHP	SW+SHP		
SAW+	SAW+		
SW+DIST	SW+DIST		
LPCLIP	LPCLIP		
SINE+	SINE+		
NOISE+	NOISE+		

Algorithm:107



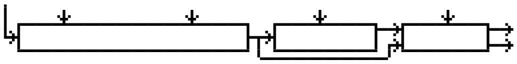
NONE	PANNER	AMP
LOPAS2 GAIN		
HIPAS2 GAIN		
BAND2 GAIN		
NOTCH2 GAIN		
LP2RES GAIN		
SHAPE2 GAIN		
LPGATE GAIN		
PARA MID		

Algorithm:108



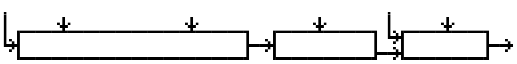
NONE	NONE	BAL AMP
AMP	AMP	AMPU AMPL
LOPASS	LOPASS	
HIPASS	HIPASS	
ALPASS	ALPASS	
GAIN	GAIN	
SHAPER	SHAPER	
DIST	DIST	
SINE	SINE	
LFSIN	LFSIN	
SW+SHP	SW+SHP	
SAW+	SAW+	
SW+DIST	SW+DIST	
LPCLIP	LPCLIP	
SINE+	SINE+	
NOISE+	NOISE+	

Algorithm:109



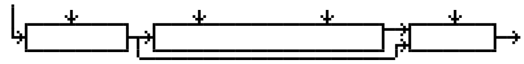
NONE	NONE	xAMP
LOPAS2 GAIN	AMP	+AMP
HIPAS2 GAIN	LOPASS	!AMP
BAND2 GAIN	HIPASS	
NOTCH2 GAIN	ALPASS	
LP2RES GAIN	GAIN	
SHAPE2 GAIN	SHAPER	
LPGATE GAIN	DIST	
PARA MID	SINE	
	LFSIN	
	SW+SHP	
	SAW+	
	SW+DIST	
	LPCLIP	
	SINE+	
	NOISE+	

Algorithm:110



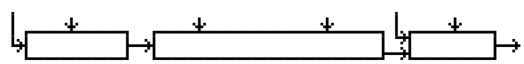
NONE	NONE	xAMP
LOPAS2 GAIN	AMP	+AMP
HIPAS2 GAIN	LOPASS	!AMP
BAND2 GAIN	HIPASS	
NOTCH2 GAIN	ALPASS	
LP2RES GAIN	GAIN	
SHAPE2 GAIN	SHAPER	
LPGATE GAIN	DIST	
PARA MID	SINE	
	LFSIN	
	SW+SHP	
	SAW+	
	SW+DIST	
	LPCLIP	
	SINE+	
	NOISE+	

Algorithm:111



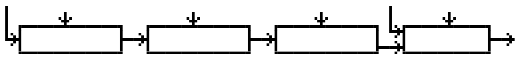
NONE	NONE	xAMP
AMP	LOPAS2 GAIN	+AMP
LOPASS	HIPAS2 GAIN	!AMP
HIPASS	BAND2 GAIN	
ALPASS	NOTCH2 GAIN	
GAIN	LP2RES GAIN	
SHAPER	SHAPE2 GAIN	
DIST	LPGATE GAIN	
SINE	PARA MID	
LFSIN		
SW+SHP		
SAW+		
SW+DIST		
LPCLIP		
SINE+		
NOISE+		

Algorithm:112



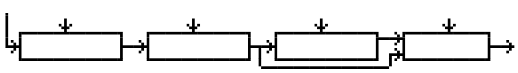
NONE	NONE	xAMP
AMP	LOPAS2 GAIN	+AMP
LOPASS	LP2RES GAIN	!AMP
HIPASS	SHAPE2 GAIN	
ALPASS	LPGATE GAIN	
GAIN	NOISE+ GAIN	
SHAPER		
DIST		
SINE		
LFSIN		
SW+SHP		
SAW+		
SW+DIST		
LPCLIP		
SINE+		
NOISE+		

Algorithm:113



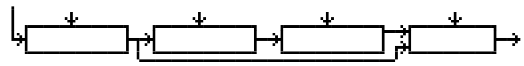
NONE	NONE	NONE	xAMP
AMP	AMP	AMP	+AMP
LOPASS	LOPASS	LOPASS	!AMP
HIPASS	HIPASS	HIPASS	
ALPASS	ALPASS	ALPASS	
GAIN	GAIN	GAIN	
SHAPER	SHAPER	SHAPER	
DIST	DIST	DIST	
SINE	SINE	SINE	
LFSIN	LFSIN	LFSIN	
SW+SHP	SW+SHP	SW+SHP	
SAW+	SAW+	SAW+	
SW+DIST	SW+DIST	SW+DIST	
LPCLIP	LPCLIP	LPCLIP	
SINE+	SINE+	SINE+	
NOISE+	NOISE+	NOISE+	

Algorithm:114



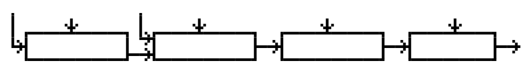
NONE	NONE	NONE	xAMP
AMP	AMP	AMP	+AMP
LOPASS	LOPASS	LOPASS	!AMP
HIPASS	HIPASS	HIPASS	
ALPASS	ALPASS	ALPASS	
GAIN	GAIN	GAIN	
SHAPER	SHAPER	SHAPER	
DIST	DIST	DIST	
SINE	SINE	SINE	
LFSIN	LFSIN	LFSIN	
SW+SHP	SW+SHP	SW+SHP	
SAW+	SAW+	SAW+	
SW+DIST	SW+DIST	SW+DIST	
LPCLIP	LPCLIP	LPCLIP	
SINE+	SINE+	SINE+	
NOISE+	NOISE+	NOISE+	

Algorithm:115



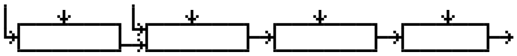
NONE	NONE	NONE	xAMP
AMP	AMP	AMP	+AMP
LOPASS	LOPASS	LOPASS	!AMP
HIPASS	HIPASS	HIPASS	
ALPASS	ALPASS	ALPASS	
GAIN	GAIN	GAIN	
SHAPER	SHAPER	SHAPER	
DIST	DIST	DIST	
SINE	SINE	SINE	
LFSIN	LFSIN	LFSIN	
SW+SHP	SW+SHP	SW+SHP	
SAW+	SAW+	SAW+	
SW+DIST	SW+DIST	SW+DIST	
LPCLIP	LPCLIP	LPCLIP	
SINE+	SINE+	SINE+	
NOISE+	NOISE+	NOISE+	

Algorithm:116



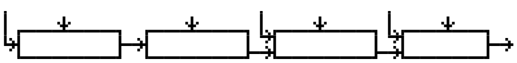
NONE	NONE	NONE	AMP
AMP	xGAIN	AMP	
LOPASS	+GAIN	LOPASS	
HIPASS	!GAIN	HIPASS	
ALPASS		ALPASS	
GAIN		GAIN	
SHAPER		SHAPER	
DIST		DIST	
SINE		SINE	
LFSIN		LFSIN	
SW+SHP		SW+SHP	
SAW+		SAW+	
SW+DIST		SW+DIST	
LPCLIP		LPCLIP	
SINE+		SINE+	
NOISE+		NOISE+	

Algorithm:117



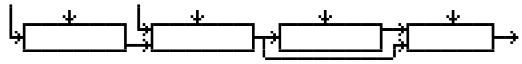
NONE	NONE	NONE	AMP
AMP	xGAIN	LOPAS2	
LOPASS	+GAIN	HIPAS2	
HIPASS	!GAIN	LPGATE	
ALPASS		LP2RES	
GAIN		SHAPE2	
SHAPER			
DIST			
SINE			
LFSIN			
SW+SHP			
SAW+			
SW+DIST			
LPCLIP			
SINE+			
NOISE+			

Algorithm:118



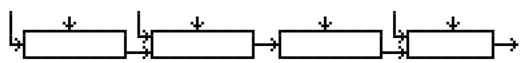
NONE	NONE	NONE	xAMP
AMP	AMP	xGAIN	+AMP
LOPASS	LOPASS	+GAIN	!AMP
HIPASS	HIPASS	!GAIN	
ALPASS	ALPASS		
GAIN	GAIN		
SHAPER	SHAPER		
DIST	DIST		
SINE	SINE		
LFSIN	LFSIN		
SW+SHP	SW+SHP		
SAW+	SAW+		
SW+DIST	SW+DIST		
LPCLIP	LPCLIP		
SINE+	SINE+		
NOISE+	NOISE+		

Algorithm:119



NONE	NONE	NONE	xAMP
AMP	xGAIN	AMP	+AMP
LOPASS	+GAIN	LOPASS	!AMP
HIPASS	!GAIN	HIPASS	
ALPASS		ALPASS	
GAIN		GAIN	
SHAPER		SHAPER	
DIST		DIST	
PWM		SINE	
SINE		LFSIN	
LF SIN		SW+SHP	
SW+SHP		SAW+	
SAW+		SW+DIST	
SAW		LPCLIP	
LF SAW		SINE+	
SQUARE		NOISE+	
LF SQR			
WRAP			

Algorithm:120

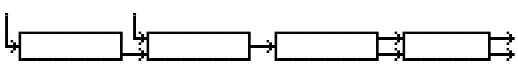


NONE	NONE	NONE	xAMP
AMP	xGAIN	AMP	+AMP
LOPASS	+GAIN	LOPASS	!AMP
HIPASS	!GAIN	HIPASS	
ALPASS		ALPASS	
GAIN		GAIN	
SHAPER		SHAPER	
DIST		DIST	
SINE		SINE	
LFSIN		LFSIN	
SW+SHP		SW+SHP	
SAW+		SAW+	
SW+DIST		SW+DIST	
LPCLIP		LPCLIP	
SINE+		SINE+	
NOISE+		NOISE+	

Algorithm:121

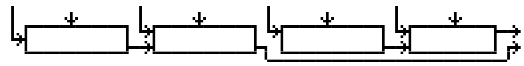
NONE	NONE	NONE	xAMP
AMP	xGAIN	xGAIN	+AMP
LOPASS	+GAIN	+GAIN	!AMP
HIPASS	!GAIN	!GAIN	
ALPASS			
GAIN			
SHAPER			
DIST			
SINE			
LFSIN			
SW+SHP			
SAW+			
SW+DIST			
LPCLIP			
SINE+			
NOISE+			

Algorithm:122



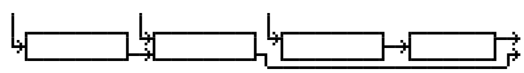
NONE	NONE	PANNER	AMP
AMP	xGAIN		
LOPASS	+GAIN		
HIPASS	!GAIN		
ALPASS			
GAIN			
SHAPER			
DIST			
SINE			
LFSIN			
SW+SHP			
SAW+			
SW+DIST			
LPCLIP			
SINE+			
NOISE+			

Algorithm:123



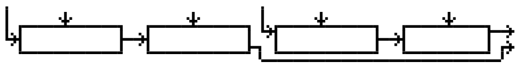
NONE	xAMP	NONE	xAMP
AMP	+AMP	AMP	+AMP
LOPASS	!AMP	LOPASS	!AMP
HIPASS		HIPASS	
ALPASS		ALPASS	
GAIN		GAIN	
SHAPER		SHAPER	
DIST		DIST	
SINE		PWM	
LFSIN		SINE	
SW+SHP		LFSIN	
SAW+		SW+SHP	
SW+DIST		SAW+	
LPCLIP		SW+DIST	
SINE+		LPCLIP	
NOISE+		SINE+	
		NOISE+	

Algorithm:124



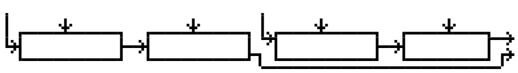
NONE	xAMP	NONE	AMP
AMP	+AMP	LOPAS2	
LOPASS	!AMP	HIPAS2	
HIPASS		LPGATE	
ALPASS		LP2RES	
GAIN		SHAPE2	
SHAPER			
DIST			
SINE			
LFSIN			
SW+SHP			
SAW+			
SW+DIST			
LPCLIP			
SINE+			
NOISE+			

Algorithm:125



NONE	AMP	NONE	AMP
AMP		AMP	
LOPASS		LOPASS	
HIPASS		HIPASS	
ALPASS		ALPASS	
GAIN		GAIN	
SHAPER		SHAPER	
DIST		DIST	
SINE		SINE	
LFSIN		LFSIN	
SW+SHP		SW+SHP	
SAW+		SAW+	
SW+DIST		SW+DIST	
LPCLIP		LPCLIP	
SINE+		SINE+	
NOISE+		NOISE+	

Algorithm:126



NONE	AMP	NONE	AMP
LOPAS2		LOPAS2	
HIPAS2		HIPAS2	
LPGATE		LPGATE	
LP2RES		LP2RES	
SHAPE2		SHAPE2	

K2661 Triple Programs: Controller Assignments

The following tables describe the controller assignments for the 10 ROM-base triple programs provided with v2. Table 12-1 lists a series of MIDI Controller numbers and their default and/or generic functions within triples. Table 12-2 lists each triple program and the specific assignments for each MIDI controller number in Table 12-1.

MIDI Controller Number	Default Function
MWheel MIDI01)	Pitch Modulation
Data (MIDI06)	Filter Attack Time
MIDI22	Filter Decay Time
MIDI23	Filter Frequency
MIDI24	Amplitude Release Time
MIDI25	Filter Resonance
MIDI26	LFO Depth (filter layers1–3, PWM layers 4–6)
MIDI27	Delay Wet/Dry
MIDI28	Reverb Wet/Dry
MIDI29	Switches Saws to PWM oscillator

Table 12-1 Default controller functions for triple programs

Program ID	Program Name	MIDI Controller	Assignment
730	Shores of Tripoli	MWheel	Sawtooth wave pitch
		Data	Lowpass filter frequency
		MIDI22	LP filter resonance (max --> min)
		MIDI23	Crossfade: partials / sawtooth
		MIDI24	Lowpass filter frequency (post-Saw)
		MIDI25	Shaper amount
		MIDI26	Resonant lowpass filter frequency
		MIDI27	Bass rolloff, treble boost (via FX)
		MIDI28	Reverb wet/dry & reverb time
		MIDI29	Attack envelope control

Table 12-2 Controller assignments for v2 factory triple programs

Program ID	Program Name	MIDI Controller	Assignment
731	Mono Triple Lead	MWheel	Pitch modulation
		Data	Shaper amount
		MIDI22	Lowpass filter frequency
		MIDI23	X gain amount
		MIDI24	Impact
		MIDI25	Chorus-delay-reverb wet/dry amount
		MIDI26	Chorus amount
		MIDI27	Delay amount
		MIDI28	Reverb time
		MIDI29	Reverb switch (Aux /insert reverb)
		MPress	Pitch modulation
732	Triple Play	MWheel	Pitch modulation
		Data	LFO on/off, filter frequency
		MIDI22	Filter resonance
		MIDI23	Pitch (Saw 2)
		MIDI24	AmpMod amount
		MIDI25	Notch2 frequency
		MIDI26	Notch2 amplitude
		MIDI27	Reverb wet/dry
		MIDI28	Pitch detune
		MIDI29	Delay mix
		MPress	Pitch modulation
733	ABCDE = ADSR+Res	MWheel	Pitch modulation
		Data	Filter attack time
		MIDI22	Filter decay time
		MIDI23	Filter frequency
		MIDI24	Amplitude release time
		MIDI25	Filter resonance
		MIDI26	LFO depth (filter layers1-3, PWM layers 4-6)
		MIDI27	Delay wet/dry
		MIDI28	Reverb wet/dry
		MIDI29	Switches saws to PWM oscillator

Table 12-2 Controller assignments for v2 factory triple programs

Program ID	Program Name	MIDI Controller	Assignment
734	StringMod Pad	MWheel	Modulating sawtooth, flange rate (FX)
		Data	Filter frequency
		MIDI22	Amplitude modulation depth
		MIDI23	Attack envelope control
		MIDI24	Frizzle distortion
		MIDI25	Aux reverb send & aux reverb time
		MIDI26	Bus2 reverb wet/dry
		MIDI27	Flange feedback
		MIDI28	Aux Reverb wet/dry & attenuation
		MIDI29	Flange rate shift
735	Plucky Emu	MWheel	Vibrato
		Data	Hi-freq stimulation / lowpass filter frequency
		MIDI22	Shaper & filter frequency / resonance
		MIDI23	Decay envelope
		MIDI24	Release envelope
		MIDI25	Aux reverb send
		MIDI26	Reverb decay time
		MIDI27	Bus2 Reverb wet/dry, stereo delay level
		MIDI28	FX2 input width, treble attenuation
736	Mad Three-Oh	MWheel	Pitch modulation
		Data	Filter frequency
		MIDI22	Filter resonance, separation, distortion, hipass gain
		MIDI23	Envelope filter frequency
		MIDI24	Filter envelope decay
		MIDI25	XFade saw/square
		MIDI26	LFO rate (for pitch modulation)
		MIDI27	(Aux) Plate reverb amount
		MIDI28	Delay mix
		MIDI29	Flange mix, insert reverb wet/dry
		MPress	Pitch modulation
737	Dastardly Drums	Data	Filters, crossfades
		MIDI22	Lowpass filters
		MIDI23	Various boosts, cuts, filter control
		MIDI24	Drum-piano intermodulation
		MIDI25	Aux reverb sends
		MIDI26	Hollow Laserverb effect
		MIDI27	Aux reverb time & HF damping
		MIDI28	FX distortion Level
		MIDI29	Switch hi-hats to shaped noise waveform

Table 12-2 Controller assignments for v2 factory triple programs

Program ID	Program Name	MIDI Controller	Assignment
738	H.Sync Rhythm	MWheel	Clocked pitch effect depth
		Data	Lowpass frequency
		MIDI22	Lowpass2 freq / Parametric EQ frequency
		MIDI23	Lowpass2 Gain / Shaper Amount
		MIDI24	Crossfade master / slave
		MIDI25	Stereo delay level
		MIDI26	Stereo delay feedback
		MIDI27	Chorus depth
		MIDI28	Chorus feedback
		MIDI29	Switch to alternative program

Table 12-2 Controller assignments for v2 factory triple programs

Alphanumeric Buttonpad Entries for DSP Functions

The tables on the following pages list the entries that you can use on the K2661's alphanumeric buttonpad to select DSP functions without having to scroll through a long list of values. Table 12-3 is sorted alphabetically by the name of the DSP function, and

When you're on the ALG page, use the cursor buttons to select the DSP block whose function you want to change, then type the numeric value corresponding to the function you want to select. Press **Enter** after typing the value. For example, to select a DSP function of **None** in a single-block function, type **6**, **0**, **Enter**.

Special Cases

In some algorithms, there are nonstandard DSP blocks containing specialized functions. These functions have different numeric entries, as listed below. In all other cases, numeric entries select DSP functions as listed in Table 12-3 and Table 12-4.

Algorithm and DSP Block	Numeric Entry	DSP Function
F1 in Algorithm 126	105	xAMP
	106	+AMP
	107	!AMP
F2 in Algorithms 123 and 124	108	LP2RES
	109	SHAPE2
	110	BAND2
	111	NOTCH2
	112	LOPAS2
	113	HIPAS2
	114	LPGATE

Function Block Size	DSP Function	Numeric Entry
Single-block	!AMP	75
	+AMP	49
	+GAIN	42
	ALPASS	17
	AMP	110
	AMPMOD	44
	BAND2	35
	DIST	20
	GAIN	18
	HIPAS2	52
	HIPASS	16
	LF SAW	28
	LF SINE	24
	LF SQR	30
	LOPAS2	37
	LOPASS	15
	LP2RES	73
	LPCLIP	70
	LPGATE	57
	MASTER	77
	NOISE+	76
	NONE	60
	NOTCH2	36
	PWM	22
	SAW	27
	SAW+	26
	SHAPE2	74
	SHAPER	19
	SINE	23
	SINE+	71
	SLAVE	78
	SQUARE	29
	SW+DST	53
	SW+SHP	25
	WRAP	31
	XFADE	43
	xAMP	48
	xGAIN	41

Table 12-3 Buttonpad entries for DSP functions (sorted alphabetically)

Function Block Size	DSP Function	Numeric Entry
Two-block	+ SHAPE MOD OSC	67
	2PARAM SHAPER	64
	2POLE ALLPASS	5
	2POLE LOPAS	2
	AMP MOD OSC	72
	AMP U AMP L	38
	BAL AMP	39
	BAND2 GAIN	85
	BANDPASS FILT	3
	HIPAS2 GAIN	84
	LOPAS2 GAIN	83
	LPGATE GAIN	89
	LPRES2 GAIN	87
	NONE	61
	NOTCH FILT	4
	NOTCH2 GAIN	86
	PARA BASS	8
	PARA MID	51
	PARA TREBLE	9
	SHAPE MOD OSC	68
	SHAPE2 GAIN	88
	x SHAPE MOD OSC	66
Three-block	4 POLE HIPASS W/SEP	54
	4 POLE LOPASS W/SEP	50
	DOUBLE NOTCH W/SEP	56
	HIFREQ STIMULATOR	44
	NONE	62
	PARAMETRIC EQ	13
	STEEP RESONANT BASS	14
	TWIN PEAKS BANDPASS	55

Table 12-3 Buttonpad entries for DSP functions (sorted alphabetically)

Function Block Size	Numeric Entry	DSP Function
Single-block	15	LOPASS
	16	HIPASS
	17	ALPASS
	18	GAIN
	19	SHAPER
	20	DIST
	22	PWM
	23	SINE
	24	LF SINE
	25	SW+SHP
	26	SAW+
	27	SAW
	28	LF SAW
	29	SQUARE
	30	LF SQR
	31	WRAP
	35	BAND2
	36	NOTCH2
	37	LOPAS2
	41	xGAIN
	42	+GAIN
	43	XFADE
	44	AMPMOD
	48	xAMP
	49	+AMP
	52	HIPAS2
	53	SW+DST
	57	LPGATE
	60	NONE
	70	LPCLIP
	71	SINE+
	73	LP2RES
	74	SHAPE2
	75	!AMP
	76	NOISE+
	77	MASTER
	78	SLAVE
	110	AMP

Table 12-4 Buttonpad entries for DSP functions (sorted numerically)

Function Block Size	Numeric Entry	DSP Function
Two-block	2	2POLE LOPAS
	3	BANDPASS FILT
	4	NOTCH FILT
	5	2POLE ALLPASS
	8	PARA BASS
	9	PARA TREBLE
	38	AMP U AMP L
	39	BAL AMP
	51	PARA MID
	61	NONE
	64	2PARAM SHAPER
	66	x SHAPE MOD OSC
	67	+ SHAPE MOD OSC
	68	SHAPE MOD OSC
	72	AMP MOD OSC
	83	LOPAS2 GAIN
	84	HIPAS2 GAIN
	85	BAND2 GAIN
	86	NOTCH2 GAIN
	87	LPRES2 GAIN
	88	SHAPE2 GAIN
	89	LPGATE GAIN
Three-block	13	PARAMETRIC EQ
	14	STEEP RESONANT BASS
	44	HIFREQ STIMULATOR
	50	4 POLE LOPASS W/SEP
	54	4 POLE HIPASS W/SEP
	55	TWIN PEAKS BANDPASS
	56	DOUBLE NOTCH W/SEP
	62	NONE

Table 12-4 Buttonpad entries for DSP functions (sorted numerically)

Appendix A Specifications

K2661 Features

- 61 note synth action keyboard with aftertouch
- 240 x 64-pixel backlit fluorescent graphic display with adjustable contrast and brightness
- Power effects processor with 4 insert effects and 1 aux effect.
- SmartMedia card slot for SmartMedia cards 4M and larger
- MIDI In, Thru, and Out with selectable second MIDI Out
- 48-note polyphony with dynamic voice allocation
- Multi-timbral, for multi-track sequencing and recording
- More than 500 factory preset programs, and more than 100 factory preset setups
- Up to 32 layers per program
- 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
- 16-bit sample ROM, including acoustic instrumental sounds, waveforms, and noise
- 20 KHz maximum bandwidth
- Optional stereo sampler with analog and digital inputs
- ADAT digital input
- Digital optical output switchable for ADAT, AES/EBU, or S/PDIF format
- Sound ROM expandable to a total of 28 Megabytes
- 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)
- Four 1/4-inch balanced audio outputs programmable as two stereo pairs or as four separate outputs
- Stereo headphone jack

- 1500K battery-backed RAM for user programs, setups and other objects
- One SCSI port for connection with an external SCSI disk, CD-ROM drive, or personal computer
- 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. Additional algorithms available for Triple Mode.
- Filters: Lowpass, Highpass, Allpass, Bandpass, Notch, programmable resonance
- Programmable stereo multi-effects on balanced 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	K2661	
Length	39.4 in	100.2 cm
Width	14.2 in	36 cm
Height	4.3 in	11 cm
Weight	35 lb	15.86 Kg

Electrical Specifications

AC supply: selectable; 100V, 120V, 230V, or 240V. 1.0 amps at 120 volts nominal

Safe Voltage Ranges

Voltage setting:	100V	120V	230V	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 K2661 will reset, but no data will be lost. If the voltage exceeds the maximum safe level, the K2661 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: K2661

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 K2661 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 K2661'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 K2661.

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 K2661s, use a different ID value for each one
78	120	Product ID (78 is K2000/K2500/K2600/K2661)
1B	27	Message type (1B is KDFX control)

Every KDFX-control SysEx message you send to the K2661 must start with this string of numerals. This lets the K2661 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 K2661 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 K2661 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 K2661 ROM Objects

The preset programs in the K2661 are organized by instrument category. You'll find a few representatives of each instrument sampled, 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.

Groove Setups

Setups 1–30 are Groove Setups. Once you've installed the objects, you can access the setups by pressing the **Setup** button on the front panel of your K2661.

When you are playing a Groove Setup, you can activate a drum pattern (actually a song file) by pressing any key below C3 (C below middle C). Once triggered, the drum pattern is automatically held or latched (in other words, you do not need to keep holding the key down for the groove to continue playing). Most grooves have a bass sound assigned to the left hand keyboard region, as well as some sounds for right hand playing.

Use your K2661's large ribbon to activate a fill for the groove. (There is one groove that does not follow this convention, #2, where there is no fill on the ribbon. Instead, a 'toms fill' is activated when you play between C3 and C4 on the keyboard.)

Note: After pressing **panic**, grooves won't trigger; you must scroll away and then back for the setup to get the correct entry value.

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 (shown below). The NewZn parameter uses this setup as its template for creating new zones.

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	

QA Banks

id	bank name
1	Pianos
2	E Pianos
3	Organs
4	Strings
5	Voices
6	Ensembles
7	Guitars 1
8	Guitars 2
9	Basses
10	Synth Basses
11	Drums 1
12	Drums 2
13	Percussion
14	Solo Brass
15	Section Brass
16	Winds
17	Analog Synths
18	Synths Leads
19	Digital Synths
20	Synth Pads
21	Synth Ambient
22	Keys
23	More Synths
24	KB3
25	Basic QA Bank

Setups

See *Groove Setups* (above) for information about Groove Setups (setups 1–30).

id	setup	long ribbon function
1	Tripped Up Fonk	Fill
2	Like Groovay	Clear Setup
3	1984 Funkhouse	Fill
4	On The Bell	Fill
5	FilteredFreak	Fill
6	MakinSweetLove	Fill
7	Tomsemble	Fill
8	Salsa-esque	Fill
9	Pickin&Grinnin	Fill
10	Funk Street	Fill
11	Rockin'Redneck	Fill
12	OldSkool SynJam	Fill
13	Progresso	Fill
14	Trio 4 Groovin	Fill
15	Fresh Tracks	Fill
16	Survival	Fill
17	SUV Ad?	Fill
18	80's LoveJam	Fill
19	Hoe Down!	Fill
20	FrEaKeD OuT	Fill
21	303/808 Madness	Fill
22	Dance Madness	Fill
23	Rave Madness	Fill
24	StrangeMixstriss	Fill
25	808Flangelicious	Fill
26	Surreal Groove	Fill
27	Hickup Groove	Fill
28	Newjack Groove	Fill
29	Nonlinear Jam	Fill
30	We Be JahMon	Fill
31	Nogorov Arp	pitch bend
32	Desert Rose	pitch bend
33	Arp Bell Pad	arp shift limit
34	Intergalactica	arp note shift
35	Flute Arps	pitch bend - flute arp layer only
36	Pad/Arp Rbn Walk	env ctl arp zone
37	Arp Bell Pad 2	delay feedback level
38	Hold & Tap	"percussion trigger, fx"
39	Aqua Ribbon	filter freq
40	Slo Wood Pad	LP Freq
41	Jazz Guitar Trio	pitch bend
42	Folk Rhythm Sect	pitch bend - bass only
43	Shades of Bombay	mark tree trigger
44	Jazz Ensemble	pitch bend
45	Stevie Bass/EP	pitch bend - bass only
46	Polar Reverie	pitch bend
47	Triple Trip	LP Freq
48	Vortex Coil	pitch bend

id	setup	long ribbon function
49	Barren Landscape	Lunar Wind trigger
50	Otherworldly	LP freq
51	Super Lush	pitch bend
52	Pad Soundscape	BP Freq
53	Glassy Eyed	pitch bend
54	Expansive	LP Freq
55	Ethereal Shadows	flanger feedback level
56	Sparkle & Bass	pitch bend
57	Vintage Poly	pitch bend
58	Big Analog	LP Resonance
59	Searing Lead	pitch bend
60	Poly Pitcher	pitch bend
61	Liquid Guitars	pitch bend
62	Roto 12 String	pitch bend
63	Nylon & Steel	pitch bend
64	Layered Guitars	pitch bend
65	We're Plucked	pitch bend
66	Cathedral	pitch bend
67	RbnSpltB3+MIDIPd	Splits (via zone mutes)
68	Registrations	pitch bend
69	Pipes & Choir	pitch bend
70	Elegant Grandeur	pitch bend
71	Cinematic Strngs	pitch bend
72	Chamber Players	pitch bend
73	18th Century	pitch bend
74	Harp/Fl & Str	pitch bend
75	Tutti Orch	pitch bend
76	Chorused Piano	pitch bend
77	Funky Keys	pitch bend
78	Piano & Vibes	pitch bend
79	FM & Tines EP	pitch bend
80	Ballad Keys	pitch bend
81	Gnu Age Piano	pitch bend
82	Digi Keys	pitch bend
83	FM & Tines EP 2	pitch bend
84	Big Key Stack	pitch bend
85	Dynamic Stack	pitch bend
86	Organ/Synth Solo	pitch bend - synth lead only
87	Guitar / Flute	pitch bend
88	Puffy Winds	pitch bend
89	Real & Syn Str	pitch bend
90	Ruggratts	pitch bend
91	Orchestral Keys	pitch bend
92	Tutti Strings	pitch bend
93	Orch Pno & Pizz	pitch bend
94	Press Roll Timps	pitch bend
95	Dreamy Fairlite	Filter Freq
96	Pad W/ Rotor	pitch bend
97	ControlSetup	pitch bend
98	Clear Setup	none
99	Default Setup	pitch bend

Songs

id	song name
1	New Song
2	Tripped Up Arr
3	Tripped Up Grv
4	Tripped Up Fll
5	Groovay Grv
6	Groovay Toms
7	1984 Funk Arr
8	1984 Funk Grv
9	1984 Funk Fll
10	On The Bell Arr
11	On The Bell Grv
12	On The Bell Fll
13	Filter Freak Arr
14	Filter Freak Grv
15	Filter Freak Fll
16	MakinLove Arr
17	MakinLove Grv
18	MakinLove Fll
19	Tomsemble Arr
20	Tomsemble Grv
21	Tomsemble Fll
22	Salsa-esque Arr
23	Salsa-esque Grv
24	Salsa-esque Fll
25	Pick&Grin Arr
26	Pick&Grin Grv
27	Pick&Grin Fll
28	Funk Street Arr
29	Funk Street Grv
30	Funk Street Fll
31	RocknRedneck Arr
32	RocknRedneck Grv
33	RocknRedneck Fll
34	OldSkool Arr
35	OldSkool Grv
36	OldSkool Fll
37	Progresso Arr
38	Progresso Grv
39	Progresso Fll
40	Trio 4 Arr
41	Trio 4 Grv
42	Trio 4 Fll
43	Fresh Tracks Arr
44	Fresh Tracks Grv
45	Fresh Tracks Fll
46	Survival Arr
47	Survival Grv
48	Survival Fll
49	SUV Ad? Arr
50	SUV Ad? Grv
51	SUV Ad? Fll
52	80sLoveJam Arr
53	80sLoveJam Grv
54	80sLoveJam Fll
55	HoeDown! Arr
56	HoeDown! Grv
57	HoeDown! Fll
58	FrEaKeD OuT Arr
59	FrEaKeD OuT GrV
60	FrEaKeD OuT FlL
61	303/808 Mad Arr
62	303/808 Mad Grv
63	303/808 Mad Fll
64	DanceMadness Arr
65	DanceMadness Grv
66	DanceMadness Fll
67	Rave Madness Arr
68	Rave Madness Grv
69	Rave Madness Fll
70	StrangeMix Arr
71	StrangeMix Grv
72	StrangeMix Fll
73	808flange Arr
74	808flange Grv
75	808flange Fll
100	Surreal Arr
101	Surreal Grv
102	Surreal Fll
103	Hickup Arr
104	Hickup Grv
105	Hickup Fll
106	Newjack Arr
107	Newjack Grv
108	Newjack Fll
109	Nonlinear Arr
110	Nonlinear Grv
111	Nonlinear Fll
112	JahMon Arr
113	JahMon Grv
114	JahMon Fll
115	Groovay Arr

Programs

id	name	ctrl	function
1	Concert Piano	MIDI25	(aux) Hall Lvl+Time
		MIDI29	Soundboard W/D
		Soft Pedal is active	
2	Stereo Solo Pno	Data	InEQ: Treb
		MIDI25	(aux) Hall Lvl+Time
		MIDI29	Soundboard W/D
		Soft Pedal is active	
3	Piano & Strings	MWheel	String Balance - softer
		Data	String Balance - louder
		MIDI25	(aux) Hall Lvl+Time
		Soft Pedal is active	
4	Pno & Syn String	MWheel	String Fade
		Data	String Swell
		MIDI23	SRS Space
		MIDI25	"Room Rev Time, Wet/ Dry"
5	Rock Grand	MIDI25	(aux) Hall Lvl+Time
		MIDI29	Soundboard W/D
		Soft Pedal is active	
6	Dyn Epiano	MWheel	Tremolo/ Vibrato
		Data	Chorus LFODepth+Rate, (aux) Plate Lvl cut+PreDly adj
		MIDI22	Chorus W/D
		MIDI23	Chorus LFODepth
		MIDI24	Chorus Xcouple
		MIDI25	(aux) Plate W/D+Decay Time
		MIDI26	Plate Room Size
		MIDI27	Chorus FB
		MIDI28	Chorus Tap Lvl
		MIDI29	Chorus Rate adj
7	Studio Class EP	MWheel	Stereo Tremolo
		Data	Tremolo Rate
		MIDI22	Phaser Rate
		MIDI23	Reverb Hi Freq Dampening (Brightness)
		MIDI24	PhaserWet/Dry
		MIDI25	Reverb Wet/Dry
		MIDI26	Distortion Warmth
		MIDI27	Distortion Drive
		MIDI28	Reverb Density
		MIDI29	Lo Freq Cut
8	The Phase EP	MWheel	Enables Stereo Tremolo
		Data	Tremolo Rate
		MIDI 22	Phaser Rate
		MIDI 23	Phaser Center Freq (Tone)
		MIDI 25	Reverb Wet/Dry

id	name	ctrl	function
9	Classic FM EPno	MWheel	"LFO Detune, Layer Delay"
		Data	Tine Overtones (modulator pitch)
		MIDI22	FM Depth
		MIDI23	Attack Rate
		MIDI24	LFO Pan Depth
		MIDI25	(Aux) Hall level
		MIDI26	FX3 Rev Time, Aux Hall Time
		MIDI27	Chorus Feedback
		MIDI28	Reverb Predelay
		MIDI29	Reverb in/out
10	Funk Clav	MWheel	Vibrato
		Data	Defeat release layer
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) HF Damping
		MIDI27	Compression Ratio & MakeUpGain
		MIDI28	(Aux) Pre-Delay
11	VAST B3	MWheel	Leslie Depth
		Data	Drawbar 1
		MIDI22	Drawbar 2
		MIDI23	Drawbar 3
		MIDI24	"Drawbar 4, EnvCtl: Imp"
		MIDI25	"Drawbar 5,6"
		MIDI26	Drawbar 7
		MIDI27	Drawbar 8
		MIDI28	Drawbar 9
		MIDI29	toggle: Vib/Chorus I/O
		Breath	"(aux) Plate Lvl, Dist Drive+adj, EQ Bass+Treb"
12	Gospel Organ	MWheel	Leslie Depth
		Data	Drawbar 1
		MIDI22	Drawbar 2
		MIDI23	"Drawbar 3, (aux) Plate Lvl"
		MIDI24	"Drawbar 4, Plate Time"
		MIDI25	KeyClick
		MIDI26	Perc Harmonic (Hi/Low)
		MIDI27	"HFDamp, Perc Decay"
		MIDI28	Cabinet Dist Drive + Lopass adj
		MIDI29	toggle: VibeChorus I/O

id	name	ctrl	function
13	Overdrive Organ	MWheel	Leslie Depth
		Data	Drawbar 1
		MIDI22	Drawbar 2
		MIDI23	"Drawbar 3, (aux) Plate Lvl"
		MIDI24	"Drawbar 4, Plate Time"
		MIDI25	KeyClick
		MIDI26	Perc Harmonic (Hi/Low)
		MIDI27	"HFDamp, Perc Decay"
		MIDI28	Cabinet Dist Drive+Lopass adj
		MIDI29	toggle: VibeChorus I/O
14	Chorus Organ	MWheel	Leslie depth
		Data	Timbre
		MIDI22	Vibrato/Chorus
		MIDI25	Reverb Time
		MIDI26	Trem Rate
		MIDI27	HF Damping
		MIDI29	Percussion
15	Chapel Organ	MWheel	Layer Detune
		Data	Switch Organ Stops
		MIDI22	All Pass Freq
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treble
		MIDI25	(Aux) Hall Level
		MIDI26	"FX1, (Aux) Size Scale"
		MIDI27	"FX1, (Aux) HF Damping"
		MIDI28	"FX1, (Aux) Pre-Delay"
16	Fast Strings	MWheel	Low pass filter cutoff (duller)
		MIDI25	Reverb Wet/Dry
		MIDI26	Reverb Time
		MIDI29	toggle: Room Ambience
17	Ster Slo Strings	MWheel	Lo Pass Res Filter Cut Off (duller)
		Data	Lo Pass non res filter Cut Off (duller)
		MIDI22	Lo Pass Res Filter Cut Off (Brighter)
		MIDI23	Env Atk Ctl
		MIDI24	Env Release Ctl
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Hall Rev Time
		MIDI27	FX1 Reverb Wet/Dry (dryer)
		MIDI28	FX1 Reverb Time (shorter)
18	Solo Arco Violin	MWheel	Envelope Attack Rate
		Data	Low pass filter cutoff (duller)
		MIDI25	(Aux) Hall Level
		MIDI26	(Fx1) Room Wet/Dry (dryer)
		MPress	"Vibrato Rate, Depth"
19	Pachelbel Strngs	MWheel	Fade Solo Strings
		Data	Fade Ensemble Strings
		MIDI25	(Aux) Rev Time (ensemble strings)
		MIDI26	(Fx1) Rev Time (solo strings)
20	Grand Strings	MWheel	Sweeping Notch
		Data	Timbre (duller)
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Rev Time
21	Cathedral Voices	MWheel	Timbre (brightness)
		Data	Enables Octave Layer
		MIDI25	(Aux) Hall Level
		MIDI26	"(Aux, FX1) rev time"
22	Unearthly Vox	MWheel	Slow Vibrato depth
		Data	Low Pass Cutoff
		MIDI22	Xfade
		MIDI23	Panning
		MIDI25	(FX1) Room Wet/Dry
		MIDI26	(Aux) Hall Level
23	Air Voices	MWheel	Slow Vibrato Depth
		Data	Bandpass Center Freq
		MIDI22	Bandpass Width
		MIDI25	(Aux) Wet/Dry (dryer)
		MIDI26	"(Aux) HF Damping, Bass Roll-off"
		MIDI27	(Aux) Reverb Time
		MIDI28	(Aux) Treble Shelf Freq
24	Cath-drVox^8veVox	MW	"Vibrato+Rate (CathV), Sin Tremolo Rate (8veV)"
		Data	toggle: CathedralVox ^ 8veVox
		MIDI22	"EnvCtl: Att, LoPass Freq, Xfade Lo/Hi Vox(8veV)"
		MIDI23	"EnvCtl: Rel, Panner pos, 8ve jump(CathV)"
		MIDI24	InEQ: Treb cut
		MIDI25	(aux) Hall Lvl
		MIDI26	(aux) Hall Time+build Time
		MIDI27	Delay Mix+FB
		MIDI28	Flange Mix+FB
		MPress	"Vibrato+Rate (CathV), Sin Tremolo Rate (8veV)"

id	name	ctrl	function
25	Choir Strings	Data	LoPass Freq cut+Res (string)
		MIDI22	LoPass Freq cut (vox)
		MIDI23	"Lyr detune, LoPass Res"
		MIDI24	Panner Width
		MIDI25	(aux) Room Lvl
		MIDI26	(aux) Room Time
		MIDI27	Flange Lvl
		MIDI28	Flange Tempo
		MIDI29	"toggle: Room + Flange (string), ChHall + Hall (vox)"
		Mpress	InEQ Bass & Treble
26	Aaron's Finale	MWheel	defeats vel. Crash
		Data	Layer Xfade Timpani and Orch Bass Drum
		MIDI22	Fade Octave String Layer
		MIDI23	Fade Trumpet Layer
		MIDI25	Reverb Time (all reverbs)
		MIDI26	(Aux) Wet/Dry
27	Fiery Orchestra	MWheel	defeats vel. Crash
		Data	Layer Xfade Timpani and Orch Bass Drum
		MIDI22	Fade Octave String Layer
		MIDI23	Fade Octave Brass Layer
		MIDI25	(Aux) Hall Level
		MIDI26	(FX1) Rev Time
28	Total Cntrl Orch3	MWheel	defeats vel. Crash
		Data	Swaps Fr Horns for Trumpets
		MIDI25	(Aux) Hall Level
		MIDI26	Reverb Time (all verbs)
		Mpress	Swell
29	Jazz Band	MW	Tremolo (guitars)
		Data	toggle: Guitars + Horns
		MIDI22	toggle: Band and Drums
		MIDI23	Tremolo Rate
		MIDI25	"(aux) rvb Lvls, W/D"
		MIDI26	SRS Parameters (guitar Lyrs)
		MIDI27	(aux) rvb Times
		MIDI28	Early reflection Lvl, Late Lvl cut
30	Rock Trio	MW	Leslie Depth
		Data	Defeats Ride Cymbal
		MIDI22	Vibrato/Chorus
		MIDI23	Swap Guitar for Organ
		Mpress	Pitch Bend on Guitar Layer

id	name	ctrl	function
31	Steel Str Guitar	MWheel	Vibrato
		Data	Lyr Enable
		MIDI22	EnvCtl: Imp
		MIDI23	EnvCtl: Att+Dec
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Chamber W/D
		MIDI26	Chamber Time
		MIDI27	Chamber HFDamp
		MIDI28	Comp Ratio
		MIDI29	toggle: Pitch I/O
32	12 Str Guitar	MWheel	Chorusy Vibrato
		Data	Exciter gain
		MIDI25	(Aux) Wet/Dry
		MIDI26	(Aux) Reverb Time
		MIDI27	(Aux) Compression Ratio
33	Nylon Gtr & Str	MWheel	Vibrato (Guitar)
		Data	Fade Strings
		MIDI22	(FX1) Reverb Wet/Dry
		MIDI23	(FX1) Reverb Time
		MIDI25	(Aux) Reverb Level (Guitar)
		MIDI26	(Aux) Reverb Level (Strings)
34	Jazz Archtop Gtr	MWheel	Vibrato
		Data	Defeats Release Layer
		MIDI24	(Aux) Room Pre-Delay
		Miidi25	(Aux) Room Level
		MIDI26	(Aux) Rev Time
		MIDI27	Compression MakeUp Gain
		MIDI28	Compression Ratio
35	Slow Chorus Gtr	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI22	Para EQ (VAST)
		MIDI23	Layer Detune
		MIDI24	Env Ctl (decay & release)
		MIDI25	(aux) Hall Lvl
		MIDI26	"Hall Time+HFDamp, Chorus W/D"
		MIDI27	"Enhc Lo Mix, Chorus FB"
		MIDI28	Enhc Hi Mix+Drive
		MIDI29	"toggle: Enhc + Chorus, Hall + Room"
		MPress	Vibrato
36	Tele In Room	MWheel	Vibrato
		Data	Tremolo Depth
		MIDI22	Tremolo Rate
		MIDI25	(Aux) Hall Level
		MIDI29	Toggle: Flanger

id	name	ctrl	function
37	Guitar Mutes 1^2	MWheel	Vibrato
		Data	Toggle: to Stereo Guitar Mutes
		MIDI22	Para EQ (VAST)
		MIDI25	(Aux) Reverb Wet/Dry
		MIDI26	(Aux) Reverb Time
		MIDI27	(Aux) HF Damping
		MIDI28	(Aux) Compression Ratio
		Mpress	Vibrato ^2
38	Spark Guitar	MWheel	Vibrato
		Data	HFStim adj
		MIDI22	EnvCtl: Imp+Att
		MIDI23	EnvCtl: Dec
		MIDI24	EnvCtl: Rel
		MIDI25	(fx1) Room Mix, (aux) Hall Lvl
		MIDI26	Hall PreDly+Time
		MIDI27	Delay Mix (sys)
		MIDI28	Chorus Dly
		MIDI29	Chorus FB
		MPress	Vibrato
39	Wah Crunch MWFT	MWheel	Wah wah
		Foot	Wah wah
		Data	Cabinet Type
		MIDI25	(Aux) Room Level
		MIDI27	FX2 Delay Wet/Dry (dryer)
40	Crunchy Lead	MWheel	Vibrato
		Data	Lyr Enable
		MIDI22	(KDFX)Dist Drive
		MIDI23	(KDFX)Dist Freq
		MIDI24	EnvCtl: Dec+Rel
		MIDI25	"(aux) FDR Lvl, Hall Time"
		MIDI26	Flange FB
		MIDI27	Flange Tempo
		MIDI28	Delay Mix
		MIDI29	Delay FB
		MPress	Lyr Balance
41	String Bass	MWheel	Vibrato
		Data	Ride Layer Enabled
		MIDI25	(FX1) Room Wet/Dry
		MIDI26	(Aux) Hall Level
		Mpress	Vibrato
42	Piano Trio	MWheel	Vibrato
		Data	Ride Cymbal Fade
		MIDI24	Treble EQ (KDFX)
		MIDI25	"(Aux) Hall Level, (FX1) wet/dry (dryer)"
		Mpress	Vibrato

id	name	ctrl	function
43	Warm Bass 1^2	MWheel	Vibrato
		Data	toggle: Lyrs
		MIDI22	"LoPass adj, Shaper amt, EnvCtl: Imp+Att"
		MIDI23	"EnvCtl: Imp, Para-Bass+HighPass Freq"
		MIDI24	"EnvCtl: Rel, InEQ: Bass"
		MIDI25	(aux) Room Lvl
		MIDI26	Room Absorption
		MIDI27	Comp Ratio
		MIDI28	Comp: Att+Rel Time
		MIDI29	add EQ Morph
		MPress	Vibrato
44	Pick It Bass	MWheel	Vibrato
		Data	"Shaper, Para Treble boost"
		MIDI25	(Aux) Hall Level
		MIDI27	Compression Ratio & MakeUp Gain
		MIDI29	"Switch to FX2, Eq Morph"
		Mpress	Vibrato
45	Dual Bass Guitar	MWheel	Vibrato
		Data	Enable Mute at Medium Velocities
		MIDI25	(Aux) Hall Level
		MIDI27	Compression Ratio & MakeUp Gain
		MIDI29	"Switch to FX2, Eq Morph"
		Mpress	Vibrato
46	Moogy Bass One	MWheel	Vibrato
		Data	LoPass Freq
		MIDI22	LoPass Res
		MIDI23	Env Ctl: Attack & Impact
		MIDI24	Env Ctl: Release
		MIDI25	"(aux) Chorus Lvl+W/D, (fx2) Room Cut"
		MIDI26	"(fx2)Chorus Mix, Enhc Crossover 1"
		MIDI27	"Chorus FB, Enhc Cross-over 2"
		MIDI28	"Room HFDamp, Enhc Drive adj"
		MIDI29	toggle: ChorVerb + Enhc; Enhc Lo+Mid+Hi Drive
		MPress	Vibrato

id	name	ctrl	function
47	Mono Bass	MW	Vibrato
		Data	LoPass Freq
		MIDI22	"LoPass Freq, Impact"
		MIDI23	Env Ctl: Attack
		MIDI24	Env Ctl: release
		MIDI25	(aux) CDR Lvl+Hall Time
		MIDI26	Delay Mix
		MIDI27	Phaser FB Cut
		MIDI28	"Phaser LFO Rate, Hall Mix"
		MIDI29	"Chorus-Delay Cut, Phase Notch adj"
		MPress	Vibrato
48	Tee Bee This MW	MWheel	LoPass Freq
		Data	LoPass Res
		MIDI22	EnvCtl: Imp
		MIDI23	EnvCtl: Att
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Hall Lvl+adj
		MIDI26	Chorus W/D
		MIDI27	Chorus FB
		MIDI28	Chorus Tap Pan
		MIDI29	add Enhc
		MPress	Vibrato
49	Sequenting	MWheel	Vibrato
		Data	"Low Pass Freq, Low Pass Separation, Env Decay Ctl"
		MIDI22	Low Pass Resonance
		MIDI23	Low Pass Separation
		MIDI25	(FX1) Wet/Dry (dryer)
		MIDI26	(FX1) Reverb Time
		MIDI27	(Aux) Hall Level
		MIDI28	"(FX1) HF Damping, Bass Shelf EQ"
		MPress	Vibrato
50	Trent Bass	MWheel	LPGate Freq
		Data	"Saw+Shp Pitch, Atk Ctl"
		MIDI25	(FX1) Wet/Dry (dryer)
		MIDI26	(FX1) Reverb Time
		MIDI27	(Aux) Hall Level
		MIDI28	"(FX1) HF Damping, Bass Shelf EQ"
51	2 Live Kits 2 MW	MWheel	Multiple Layer toggle
		Data	"Pitch: Kicks, Toms"
		MIDI22	Pitch: Snares
		MIDI23	"HF Stimulator: Cymbal, HiHats"
		MIDI24	"EnvCtl: Kicks, Snares, Toms, Cymbal"
		MIDI25	"(FX1)-(aux) Hall Lvl, (FX2) Plate PreDly"
		MIDI26	(FX2)-(aux) Hall Lvl
		MIDI27	(FX1) GateRvb W/D+Gate Threshold
		MIDI28	"Hall Time, Plate W/D"
		MIDI29	toggle: Plate RvrbTime boost-Megaverb!
52	Jazz Kit II	MWheel	Pitch: AuxPerc
		Data	"Pitch: Kicks, Toms"
		MIDI22	Pitch: Snares
		MIDI23	"Gain: HiHats, Crash Cym"
		MIDI24	"EnvClt: Kicks, Toms"
		MIDI25	(FX1+2) Rooms W/D+Time
		MIDI26	"(FX1+2)- (aux) Hall Lvl, (FX2)- Mix Lvl"
		MIDI27	(FX2) In EQ: Treb cut
		MIDI28	(aux) Hall TrebShlf Freq+cut
53	Retro Skins MW	MWheel	Multiple Layer toggle
		Data	Pitch: Kicks
		MIDI22	Pitch: Snares
		MIDI23	"Filter Freq: Kicks, Toms, Ride, AuxPerc "
		MIDI24	EnvCtl: Kicks+Snares
		MIDI25	(FX1+2) Rooms W/D
		MIDI26	(aux) Room W/D
		MIDI27	(aux) Compressor Attack Time
		MIDI28	(FX1) InEQ: Bass+Treb
		MIDI29	toggle: Alien Skin Effect

id	name	ctrl	function
54	Lo-Fi Vinyl Kit	MWheel	Pitch for most Needle FX and other SFX
		Data	"Pitch: Kicks, Toms, HiHats"
		MIDI22	"Pitch: Snares, Crash1"
		MIDI23	Assorted Filters: Kick, Toms, Snares, HiHats, Crashes, Ride (Resonant)
		MIDI24	"EnvClt: Kick, Toms, Snares"
		MIDI25	(FX1) Booth W/D
		MIDI26	(aux) Hall Lvl
		MIDI27	"(FX2) Pitcher W/D, (FX3) LaserVerb W/D"
		MIDI28	"(FX2) Pitcher Pitch, (FX3) LaserVerb Delay"
		MIDI29	toggle: Pitcher + LaserVerb
55	VAST Sliders 808	MWheel	"AltStart control, Impact on most elements"
		Data	"Pitch: Kicks, Toms"
		MIDI22	"Pitch: Snares, NoizeToms"
		MIDI23	"EnvCtl: Kicks, Toms"
		MIDI24	"EnvCtl: Snares, HiHats, Crash2, NoizeToms"
		MIDI25	(FX1) Hall W/D
		MIDI26	(FX4)- (aux) Room Lvl (dry at very top)
		MIDI27	"Hall Time, Room Decay Time+HFDamp"
		MIDI28	"(FX2) Flange W/D+FB, (FX3) 8-Tap W/D"
		MIDI29	"toggle: 8-Tap I/O (Sys), Room Lvl adj"
56	Perc Section	MIDI22	"(FX1, FX3) Wet/Dry"
		MIDI23	(Fx2) Wet/Dry
		MIDI24	"Reverb Time FX1, FX2"
		MIDI25	(Aux) Reverb Time
		MIDI29	"Switch FX1, FX2"
57	Touch Drums	MWheel	Vibrato
		Data	Volume
		MIDI25	"(FX1) Wet/Dry, Absorb-tion"
		MIDI27	(FX2) Quantize Wet/Dry
		MIDI28	(FX2) Headroom (less)
		MIDI29	"Switch to FX bus 2, Quan-tize/Flange"
		Mpress	"Pitch Bend, Vibrato"
58	Vibraphone	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI22	"Partial Pitches, Layer Delay"
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treble
		MIDI25	(Aux) Reverb Level
		MIDI26	"(Aux) Reverb Time, Treble Shelf Gain"
		MIDI27	Chorus Mix
		MIDI28	Chorus Depth
59	Marimbae	MWheel	"EnvCtl: Rel, Tremolo"
		Data	Fade in Percussive Layer
		MIDI22	LP / HPass freq, HFStim Drive
		MIDI23	Timbre - Duller
		MIDI25	"(aux) Hall Lvl, Room W/D"
		MIDI26	Hall+Room Times
		MIDI29	toggle: Room + Compres-sor/Hall ^ Room I/O
60	Dynamic Perc	MWheel	Switch Conga Layers
		Data	Conga Pitch when MW up
		MIDI25	"FX1,3 Wet/Dry"
		MIDI26	FX2 Wet/Dry
		MIDI27	"FX1,2 Rev Times"
		MIDI28	"(Aux) Wet/Dry, Rev Time"
		MIDI29	toggle; Reverbs FX1 & 2
61	Dynasax	MWheel	"Vibrato, LoPass Freq"
		Data	Lyr enable
		MIDI22	"Lyr AltCtl, LoPass Freq, Notch Freq, ParaTreb Freq"
		MIDI23	"Notch Width, LoPass Res, EnvCtl: Imp+Att"
		MIDI24	EnvCtl: Dec+Rel
		MIDI25	(aux) Hall Lvl
		MIDI26	Hall HFDamp+Decay Time
		MIDI27	Chorus Mix
		MIDI28	Delay (sys) Mix
		MIDI29	Hall PreDly + room size adj
		MPress	"Vibrato, LoPass Freq+Res, Shape adj"
		ChanSt	"Lyr AltCtl, EnvCtl: Rel"
62	Soft Alto	MWheel	"Vibrato, Env Ctl Atk"
		MIDI25	"FX1 Wet/Dry, Reverb Time"
		MIDI26	(Aux) Hall Level
		MPress	Vibrato Depth & Rate

id	name	ctrl	function
63	DynTrum-pet^Miles	MWheel	"swell, Vibrato"
		Data	toggle: DynTrumpet ^ Miles
		MIDI22	LoPass Freq+Res
		MIDI23	"EnvCtl: Imp, InEQ: Bass"
		MIDI24	"EnvCtl: Rel, InEQ: Treb"
		MIDI25	"(fx1) Chamb W/D, (aux) Room Lvl"
		MIDI26	Chamb + Room Times
		MIDI27	"Chamb + Room HFDamp, Dist Drive"
		MIDI28	Dist LoPass Freq
		MIDI29	toggle: Chamb + Dist
		MPress	Vibrato
64	Harmon Mute Trp	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI22	Bandpass Ctr Freq
		MIDI23	Bass Shelf EQ Gain (KDFX)
		MIDI24	Treble Shelf EQ Gain (KDFX)
		MIDI25	FX1 Wet/Dry, (Aux) Hall Level
		MIDI26	"FX1, Aux Reverb Time"
		MIDI27	"FX1, Aux HF Damping"
		MPress	Vibrato
65	French Horn	MWheel	Swell
		Data	Low Pass Freq
		MIDI22	Resonance (Sliders A&B up full = Stopped [+] Mute
		MIDI25	FX3 Room Wet/Dry
		MIDI26	(Aux) Reverb Time
		Mpress	Vibrato
66	Big Band	MWheel	LoPass adj
		MIDI25	(aux) Room W/D
		MIDI26	Room Time
		MIDI27	Room PreDly
		MIDI28	Room HFDamp
		MIDI29	Enhc I/O
		MPress	Vibrato
67	Hip Brass	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI25	FX1 Wet/Dry
		MIDI26	(Aux) Hall Level
		MIDI27	(Aux) HF Damping
		MIDI29	Sweep Filt I/O
		MPress	Swell

id	name	ctrl	function
68	Brt Saxy Section	MWheel	Vibrato
		Data	"InEQ: Bass, LoPass Freq"
		MIDI22	InEQ: Treb
		MIDI23	"EnvCtl: Imp, Att+Dec"
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Room Lvl
		MIDI26	"Room W/D + HFDamp, InEQ: Treb Freq"
		MIDI27	Dist tube Drive
		MIDI28	Dist Warmth+Tone
		MIDI29	"toggle: Dist+EQ I/O, Room type"
		MPress	Vibrato
69	Brass Fanfare	MWheel	Vibrato
		Data	Fade in French Horn layer
		MIDI25	FX1 Room Wet/Dry
		MIDI26	"(Aux) Hall Level, FX1 Reverb Time"
		MIDI27	(Aux) HF Damping
		MIDI29	toggle: Hall
		MPress	Brass Swell
70	Pesante Horns	MWheel	Vibrato
		Data	Enable and Fade in Fr Hrn Section
		MIDI25	(Aux) Hall Level
		Mpress	Swell
71	Wendy's Flute	MWheel	Low Pass Freq
		Data	toggle: Flute Variation
		MIDI25	"(aux) Hall Level, Rev Time"
		MIDI29	toggle: Hall
72	Crimson Flute	MWheel	Tremolo
		Data	HF Stimulator Drive
		MIDI22	FX1 Mix Delay
		MIDI25	FX1 Wet/Dry
		MIDI26	"(Aux) Wet/Dry, Decay Time"
		MIDI27	(Aux) Pre-Delay
		MIDI28	(Aux) HF Damping

id	name	ctrl	function
73	Horn & Flute w/ Str	MWheel	"Vibrato, LoPass sep (expression / dynamic ctl)"
		Data	toggle: Horn ^ Solo String
		MIDI22	LoPass Freq+Res cut
		MIDI23	Ens Strings Vol cut
		MIDI24	Ens Strings EnvCtl: Att
		MIDI25	(aux) Hall Lvl
		MIDI26	Hall Time
		MIDI27	(FX1) Chapel W/D
		MIDI28	Chapel Time
		MIDI29	toggle: (Lyr 3+4) Chapel+Hall, (Lyr 1) Hall+Chapel
		MPress	Ens Strings Vibrato
		SostPd	toggle: Solo Strg I/O
74	Brahms Quintet	MWheel	Vibrato/Tremolo
		Data	Fade out Pizz Basses
		MIDI22	Fade out Brass
		MIDI25	"FX1 Wet/Dry, (Aux) Reverb Level"
		MIDI26	(Aux) Reverb Time
75	Kurz'd Pipe	MWheel	Vibrato
		Data	Fade Chiff Layer
		MIDI25	(Aux) Chamber Level
		MIDI26	(Aux) Reverb Time
		MIDI29	toggle: Pitcher
		Mpress	Vibrato
76	Synth Strings	MWheel	"Vibrato, modulation"
		Data	toggle: Lyr 1 ^ Lyr 3
		MIDI22	Lyr 1 up p5th ^ Lyr 3 up 8ve
		MIDI23	EnvCtl: Att
		MIDI24	EnvCtl: Imp+Rel
		MIDI25	(aux) Plate Lvl
		MIDI26	"Chorus W/D, Dist Drive"
		MIDI27	Chorus FB
		MIDI28	Dist Bass+Treb tone
		MIDI29	toggle: Chorus + Distortion
		MPress	"Vibrato, modulation"
77	ABCD = ADSR !	MWheel	Vibrato
		Data	Filter Envelope Attack
		MIDI22	Filter Envelope Decay
		MIDI23	Envelope Sustain Level
		MIDI24	Envelope Release
		MIDI25	Reverb Wet/Dry
		MIDI26	Reverb Time
		MIDI27	Chorus Delay Wet/Dry (dryer)
78	Memorymoog	MWheel	Vibrato
		Data	"Low Pass Freq,Env Ctl Attack & Release"
		MIDI23	(Aux) Lazerverb spacing
		MIDI24	(Aux) Lazerverb Contour
		MIDI25	FX1 Hall Wet/Dry
		MIDI26	FX1 Reverb Time
		MIDI27	"(Aux) Lazerverb Level, Feedback level"
		MIDI28	(Aux) Dly Coarse
		Mpress	Vibrato
79	OB Pad	MWheel	Vibrato
		Data	LoPass Freq, EnvCtl: Att+Rel
		MIDI22	LoPass Res
		MIDI25	(aux) Plate Lvl+Time
		MIDI26	"Enhc Lo Drive+Mix, Chorus W/D "
		MIDI27	"Enhc Mid Drive, Mid Mix"
		MIDI28	"Enhc Hi Drive, Hi Mix, InEQ: Treb"
		MIDI29	toggle: Enhancer + Chorus
		MPress	Vibrato
80	TeknoBallCrusher	MWheel	Vibrato
		Data	"EnvCtl: Att, Notch Freq"
		MIDI22	saw 8ve jump (Lyr 1)
		MIDI23	EnvCtl: Impact
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Room Lvl
		MIDI26	Chorus W/D; Dist Drive cut
		MIDI27	Chorus Rate; Dist warmth cut
		MIDI28	Chorus FB; Dist cab LoPass
		MIDI29	toggle: Chorus + Distortion
		MPress	Vibrato
81	AlaZawi Take 2	MWheel	Vibrato
		Data	LoPass Freq+Res
		MIDI22	LoPass Freq cut
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treb
		MIDI25	(aux) Hall Lvl+Decay Time
		MIDI26	Hall PreDly+HFDamp
		MIDI27	Chorus W/D+Pan
		MIDI28	MDelay W/D
		MIDI29	toggle: Clean +MDelay-Chorus
		Breath	LoPass Freq+Res adj
		MPress	Vibrato

id	name	ctrl	function
82	Round Lead	MWheel	Vibrato
		Data	FM Depth (timbre)
		MIDI22	Layer Delay
		MIDI23	"Env Ctl, atk & decay"
		MIDI24	Release Rate
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Flanger Level
		MIDI27	(Aux) Delay Level
		MIDI28	(Aux) All effects on/off
		MPress	"Vibrato, FM Depth (timbre)"
83	Mono Triple Lead	MWheel	Vibrato
		Data	Shaper Gain
		MIDI22	Low Pass Freq
		MIDI23	Non-Linear Mixer Gain
		MIDI25	FX3 Wet/Dry
		MIDI26	FX3 and Aux Rev Times
		MIDI27	Chorus Mix
		MIDI28	Delay Mix
		MIDI29	(Aux) Level
		MPress	Vibrato
84	Jordan's Lead	MWheel	Vibrato
		Data	Low Pass Freq & Res
		MIDI22	Resonance Layer 2
		MIDI25	(Aux) Level and Rev Time
		MIDI26	"Delay Mix, Mid EQ"
		MIDI27	(Aux) Flanger Feedback
		MIDI28	(Aux) Flanger Tempo
		MIDI29	Distortion Drive
		MPress	Fade in Feedback Layer
85	Dist Saw Lead	MWheel	Xfade Octave Feedback, Vibrato
		Data	Low Pass Freq
		MIDI22	4P Low Pass Separation and Resonance
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Wet/Dry
		MPress	Vibrato
86	Instant Enya	MWheel	Vibrato
		Data	Lyr 1 Octave Pitch Shift
		MIDI22	Lyr 2 Low Pass Freq
		MIDI23	Bass Shelf EQ Gain (KDFX)
		MIDI24	Treble Shelf EQ Gain (KDFX)
		MIDI25	(Aux) Hall Level
		MIDI26	Chorus Delay
		MIDI27	Chorus Depth
		MIDI28	Mix Delay
		MIDI29	(Aux) Pre-Delay, Decay Time
		MPress	Vibrato

id	name	ctrl	function
87	SynKey	MWheel	Vibrato
		Data	Modulator Pitch (timbre)
		MIDI22	Layer enable
		MIDI23	"Env Ctl Atk Rate, Decay Rate"
		MIDI24	Release Rate
		MIDI25	(Aux) Hall Level, Hall Size Scale
		MIDI26	(Aux) Flanger feedback level
		MIDI27	"(Aux) Delay level, Delay Feedback leve"
		MIDI28	(Aux) Delay Time
		MIDI29	(Aux) Delay Level (off/on)
		MPress	Vibrato Rate
88	Tubular Bells	MWheel	Tremolo
		Data	Pitch
		MIDI22	Modulator Pitches
		MIDI23	Attack Rate
		MIDI24	Release Rate
		MIDI25	"(FX3) Delay amount, (FX2) Phaser wet/dry"
		MIDI26	Flanger Depth
		MIDI27	(Aux) Reverb Decay Time
		MIDI28	"(FX2, FX3) Aux send, (FX3) Wet/Dry"
		MIDI29	Toggle FX3 (Flange/decay/verb)– FX2 (Phaser)
89	Digicomp	MWheel	Vibrato
		Data	"Env Ctl: Atk Rate, Dec Rate"
		MIDI22	(FX2) Env Follower Threshold
		MIDI23	(FX2) Freq Sweep
		MIDI24	(FX2) Resonance
		MIDI25	(FX2) Filter Type
		MIDI26	(FX2) Minimum Freq
		MIDI27	(FX2) Release Rate
		MIDI28	"(FX3) Feedback Level, LF Damping"
		MIDI29	Switch FX2 (env follower) to FX3 (Spectral 4Tap
		Mpress	Vibrato
90	New Highbells	MWheel	Vibrato
		Data	Pitch (sine+)
		MIDI25	"(FX1) Wet/Dry, Rev Time"
		MIDI26	(Aux) Reverb Level
		MIDI29	toggle: FX1 (Plate) - FX2 (Flange)
		Mpress	Vibrato

id	name	ctrl	function
91	Portal	MWheel	none
		Data	High Pass Freq
		MIDI22	Saw+ Pitch
		MIDI23	LFO depth - LP Freq
		MIDI24	Resonance
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Reverb Time
		MIDI27	Flange Wet/Dry
92	Beauty Pad	MWheel	Vibrato Depth
		Data	"All Pass Freq, Lyr 2 Detune"
		MIDI22	Lyr 3 Pan Position
		MIDI25	(Aux) Wet/Dry (dryer)
		MIDI26	(Aux) Reverb Time (less)
		Mpress	Vibrato Depth
93	Amp Mod Pad	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI25	(Aux) Hall Wet/Dry (dryer)
		MIDI26	(Aux) Reverb Time
		MIDI27	(Aux) HF Damping
94	Light Mist	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI22	Pitch adj
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treb
		MIDI25	(aux) Hall Lvl
		MIDI26	Chorus Delay Time
		MIDI27	Chorus Delay Depth
		MIDI28	Delay Mix (sys)
		MIDI29	Hall Time+PreDly adj
		MPress	Vibrato
95	Soft Pad	MWheel	7 step LFO depth - pitch
		Data	Low Pass Freq
		MIDI22	Resonance
		MIDI23	4P Low Pass Separation
		MIDI24	Octave Shift Lyr 1
		MIDI25	FX1 & 2 Wet/Dry (dryer)
		MIDI26	FX2 Chorus Feedback Level
		MIDI27	FX2 LFO Depth
		MIDI28	FX2 LFO Rate
96	Eyes Wired Shut	MWheel	Vibrato Depth
		Data	ShapeModOsc Pitch
		MIDI25	FX1 Wet/Dry, FX2 Hall Level
		MIDI26	FX1 HF Damping
		MIDI27	"FX2 Frequency, Out Gain"
		MIDI28	FX2 Resonance
		MIDI29	toggle: Hall to Resonant Filter
		Mpress	Vibrato
97	Ethereal Strings	MWheel	Band Pass Freq, Width, Amplitude
		Data	Lyr enable
		MIDI22	BandPass Freq + Width - Lyr 2
		MIDI23	BandPass Width - Lyr 3
		MIDI24	InEQ: Treb
		MIDI25	(aux) Hall Lvl
		MIDI26	Hall Decay Time
		MIDI27	Flange W/D
		MIDI28	Flange FB
		MIDI29	"toggle: Flange + CDR, InEQ: Bass"
		MPress	BandPass Freq
98	Sync Waves	MWheel	Slow pitch mod Master Sync Osc
		Data	Pitch Slave Sync Osc
		MIDI22	Low Pass Freq
		MIDI23	4P Low Pass Separation
		MIDI24	Hi Pass Freq
		MIDI25	"SRS Out, (Aux) Wet/Dry (dryer)"
		MIDI26	(Aux) Reverb Time
		MIDI27	(Aux) HF Damping
		MIDI28	SRS Center Ctl
99	Tripoli 2	MWheel	Vibrato
		Data	Pitch
		MIDI22	Resonance
		MIDI23	Xfade
		MIDI24	Low Pass Freq
		MIDI25	Shaper
		MIDI26	LP2 Res Gain
		MIDI27	"Bass EQ Freq, Gain"
100	Monolith	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI25	(Aux) Wet/Dry
		MIDI26	(Aux) HF Damping
		Mpress	Pitch Layer 2
101	Soft Piano	MIDI25	(aux) Hall Lvl+Time
		Soft Pedal is active	
102	Piano for Lyrs	MIDI25	(aux) Hall Lvl
		MIDI26	Hall Time
		MIDI29	Soundboard W/D
103	Grand & Electric	MWheel	E Pno Vibrato + ParaTreb
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treb
		MIDI25	(aux) Hall Lvl
		MIDI26	Chorus W/D
		MIDI27	Chorus FB
		MIDI28	Chorus XCouple
		MIDI29	(aux) Early Ref Lvl
		Soft Ped	Softens Elec Piano

id	name	ctrl	function
104	E Grand Stack	MWheel	String Lvl
		Data	InEQ: Treb boost
		MIDI25	(aux) Room Lvl, (aux) FDR W/D
		MIDI26	Flange Mix
		MIDI27	Flange Tempo
		MIDI28	Enhc Lo/Mid Drive
		MIDI29	FDR Delay Mix adj
105	ClassicPi-ano&Vox	MWheel	Vox Lvl
		Data	Vox Balance, Piano Treb boost
		MIDI22	Vox EQ Bass
		MIDI23	"Vox EQ Treb, St Image Mix"
		MIDI25	"(aux) Hall Lvl, Room W/D"
		MIDI26	Room and Hall Times
		MIDI27	St Image In Gain
		MIDI28	St Image CenterGain
		MIDI29	Vox St Image L/R Delay
106	Brt Concert Pno	Data	InEQ: Treb
		MIDI25	(aux) Hall Lvl+Time
		MIDI29	Soundboard W/D
		Soft Pedal is active	
107	Modified Piano	Data	InEQ: Treb
		MIDI25	(aux) Hall Lvl+Time
		MIDI29	Soundboard W/D
		Soft Pedal is active	
108	Studio Grand	MIDI25	(aux) Hall Lvl+Time
		MIDI29	Soundboard Rvb Enable
		Soft Pedal is Active	
109	Orchestral Piano	MIDI25	(aux) Hall Level + Time + HF Damp (less), FX1 Wet/Dry (less)
		MIDI29	Soundboard Rvb Enable
		Soft Pedal is Active	
110	Honky-Tonk	MWheel	Tremolo
		MIDI25	(aux) Hall Time
		MIDI26	(aux) Chorus Mix
		MIDI27	Chorus FB
		MIDI28	(aux) Delay Mix
		MIDI29	Delay Time adj
111	Fonk Epno MW	MWheel	Wah Filter
		Foot	Wah Filter
		Data	Tremolo Depth
		MIDI22	Tremolo Rate
		MIDI23	Env Ctl: Atk
		MIDI24	"EnvCtl: Rel, Bass EQ (KDFX)"
		MIDI25	(aux) Hall Lvl
		MIDI26	Hall Time
		MIDI27	4Tap W/D
		MIDI28	4Tap FB
		MIDI29	4Tap I/O
112	Trip Wah Clav	MWheel	Wah Filter
		Foot	Wah Filter
		Data	Enable Release Layer
		MIDI22	Low Pass Freq
		MIDI25	"(Aux) Hall Level, Rev Time"
		MIDI26	(Aux) HF Damping
		MIDI27	Compression Ratio & MakeUpGain
		MIDI28	(Aux) Pre-Delay
113	FM E Piano	MWheel	Chorusy Vibrato
		Data	Layer 1 Pitch
		MIDI22	Modulator Pitch Lyr 2
		MIDI23	Modulator Pitch Lyr 3
		MIDI25	"(FX1) Enhancer In/Out, (FX2) Chorus Wet/Dry, (FX3) CDR Wet/Dry"
		MIDI26	Enhancer Crossover
		MIDI27	Chorus Feedback Level
		MIDI28	Chorus Depth
114	Stage EP	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI22	Low Pass Freq & Res
		MIDI23	Bass EQ Gain (KDFX)
		MIDI25	(FX1) Wet/Dry
		MIDI26	(FX1) Rev Time
		MIDI27	(FX1) HF Damping
115	Growlin' EP	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI22	Resonance
		MIDI25	"(Aux) Room Level, Wet/Dry"
		MIDI26	"(Aux) Rev Time, Size Scale"
		MIDI27	(Aux) HF Damping
		MIDI28	"(FX3) Cabinet LP, Warmth"
		MIDI29	Alt Sample Start

id	name	ctrl	function
116	Ballad Organ	MWheel	Leslie Depth
		Data	Drawbar 1
		MIDI22	Drawbar 2
		MIDI23	"Drawbar 3, (aux) Plate Lvl"
		MIDI24	"Drawbar 4, Plate Time"
		MIDI25	KeyClick
		MIDI26	Perc Harmonic (Hi/Low)
		MIDI27	"HFDamp, Perc Decay"
		MIDI28	Cabinet Dist Drive+Lopass
		MIDI29	toggle: VibeChorus I/O
117	Cookin Bee	MWheel	Leslie Depth
		Data	Distortion Drive
		MIDI22	Vibrato/Chorus
		MIDI23	"(FX2) Hi,Lo Gain"
		MIDI24	"(FX2) Hi,Lo Trem"
		MIDI25	(Aux) Plate Level
		MIDI26	(Aux) Rev Time
		MIDI27	(Aux) HF Damping
118	Dance Perc Bass	MWheel	Vibrato
		Data	Disable Layer 2
		MIDI22	"Disable Layer 3, Para EQ Width Lyr 2"
		MIDI23	"Hi Pass Separation, Para EQ"
		MIDI24	Hi Pass Resonance, Env Ctl: Atk
		MIDI25	(Aux) Hall Wet/Dry
		MIDI26	(Aux) Rev Time
		MIDI27	(Aux) HF Damping
		MIDI28	Treble Shelve Freq
119	Chiffy Pipes	MWheel	Decrescendo
		Data	LoPass Freq
		MIDI22	Key Click
		MIDI23	Vibrato
		MIDI25	(aux) Hall Lvl+W/D
		MIDI26	Hall Time
		MIDI27	Hall Early Ref Lvl
		MIDI29	"toggle: Chorus I/O, Hall HFDamp+PreDly"
120	Pipe Organ 4	Data	Subtle Pitch and LP Filter modulation
		MIDI25	(FX1) Rev Time
		MIDI26	(Aux) Rev Time
121	Marcato String Orch	MWheel	Alt Attack: switched
		Data	Enable Octave Layer
		MIDI22	Treble Shelve EQ
		MIDI23	Bass Shelve EQ
		MIDI25	Hall Wet/Dry
		MIDI26	Reverb Time
122	Adagio Strings	MWheel	none
		Data	Treble Shelf EQ
		MIDI22	Bass Shelf EQ
		MIDI25	Hall Wet/Dry
		MIDI26	Reverb Time
123	Brighter Pizz	MWheel	EQ Duller
		Data	Shaper
		MIDI25	"Hall Wet/Dry, Rev Time"
		MIDI26	HF Damping
124	Slo Solo Cello	MWheel	4P Low Pass Separation
		Data	"Low Pass Freq, Resonance"
		MIDI22	Env Ctl: Decay
		MIDI23	Env Ctl: Attack
		MIDI25	(Aux) Hall Level
		MIDI26	"(Aux) Reverb Time, HF Damping"
		MIDI27	(FX1) Wet/Dry
		MIDI28	(FX1) HF Damping
		MIDI29	"toggle: Aux off, FX1 change Room preset"
		Mpress	Increase Vibrato Depth
125	Arco Bass	MWheel	Vibrato
		Data	Env Ctl: Attack
		MIDI22	Para Bass EQ
		MIDI25	(Aux) Hall Level
		MIDI26	(FX1) Wet/Dry
		Mpress	Vibrato
126	Solo Strings	MWheel	Env Ctl: Attack
		Data	Low Pass Freq
		MIDI22	Shaper
		MIDI25	(Aux) Hall Level
		MIDI26	"(Aux) Reverb Time, HF Damping"
		MIDI27	(FX1) Wet/Dry
		Mpress	Vibrato
127	Touch Strings	MWheel	Vibrato
		Data	Env Ctl: Atk & Release
		MIDI25	FX1 Wet/Dry
128	Mixed Choir	MWheel	Vibrato+Rate
		Data	Lyr XFade
		MIDI22	"EnvCtl: Rel, Notch + ParaTreb Freq"
		MIDI23	"InEQ: Bass, ParaTreb, Notch Width"
		MIDI24	InEQ: Treb
		MIDI25	(aux) Hall Lvl
		MIDI26	Room W/D
		MIDI27	Room Time
		MIDI28	Infinite Decay on Keydown
		MIDI29	Infinite Decay
		MPress	Vibrato+Rate

id	name	ctrl	function
129	Bamboo Voices	MWheel	"Vibrato, Para EQ Freq"
		Data	Boost Vox Layer
		MIDI23	Bass EQ (KDFX)
		MIDI24	Treble EQ (KDFX)
		MIDI25	(Aux) Hall Level, FX1 Wet/Dry
		MIDI26	FX1 Rev Time
		MPress	Vibrato
130	Syn Orch Power	MWheel	Vibrato+Rate
		Data	LP2Res Freq
		MIDI22	Env Ctl: Release (faster)
		MIDI25	(Aux) Reverb Time
		MIDI26	FX2 Chorus Wet/Dry
		MIDI27	FX2 Chorus Feedback Level
		MIDI28	FX2 Chorus LFO Rate
		MIDI29	Switch to FX2 Chorus
		MPress	Vibrato+Rate
131	Strummer Guitar	MWheel	Vibrato
		Data	Enhancer Drive & Gain (less)
		MIDI25	(Aux) Reverb Wet/Dry
		MIDI26	(Aux) HF Damping
		MIDI27	(Aux) Compression Ratio
		MIDI29	Switch to FX2 Pitcher
		MPress	Vibrato
132	Blue Moods	MWheel	Slight Vibrato, String Balance
		Data	String Balance, Gtr Hi Freq Cut
		MIDI22	EnvCtl: Imp+Att
		MIDI23	EnvCtl: Dec
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Hall Lvl
		MIDI26	Hall Time+HFDamp
		MIDI27	"Enhc Lo Mix, Chorus W/D"
		MIDI28	Enhc Hi Mix+Drive, Chorus FB
		MIDI29	"toggle: Enhc + Chorus, Hall + Room"
		MPress	Vibrato
133	ES335	MWheel	Notch Filt Tremolo
		Data	Para Mid Freq
		MIDI22	"Para Mid Amp (ES335), "
		MIDI23	EnvCtl: Att
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Hall Mix
		MIDI26	"Hall HFDamp, InEQ: Bass+Treb (Abercrmbie)"
		MIDI27	Chorus Mix
		MIDI28	Delay Mix
		MIDI29	Turns off Semi-Tone Pitch Bend
		MPress	Vibrato
		PWheel	Simulates Fretboard Slide (ES335)
134	Kotolin	MWheel	Para EQ AMP
		Data	"Para EQ Freq, Width, Depth"
		MIDI22	EnvCtl: Imp
		MIDI23	EnvCtl: Att
		MIDI24	EnvCtl: Rel
		MIDI25	"(aux) Hall Lvl, (Fx3) Rvb Time"
		MIDI26	(Fx2) Phase W/D
		MIDI27	"Phase L/R LFO, (Fx3) Flange Mix"
		MIDI28	Delay Mix
		MIDI29	"Buss toggle:, Phaser LFO Rate"
		MPress	Vibrato
135	Dreamguitar	MWheel	Vibrato
		Data	Octave Pitch shift Pad layer
		MIDI22	Notch Freq
		MIDI25	"FX1 Rev Mix, (Aux) Hall Level"
		MIDI26	"(Aux) Pre-Delay, Rev Time"
		MIDI27	"FX1 Rev W/D, Delay Mix"
		MIDI28	Chorus Delay
		MIDI29	Chorus Feedback
		MPress	Vibrato
136	Hyper Guitar	MWheel	Vibrato
		Data	Enhancer Amplitude
		MIDI22	Env Ctl: Decay
		MIDI23	Treble Shelf EQ Gain
		MIDI25	(Aux) Hall Level
		MIDI26	FX2 Wet/Dry
		MIDI27	FX2 LFO Rate
		MIDI29	Toggle: effect to Stereo Image
		MPress	Vibrato

id	name	ctrl	function
137	SliderDistJazzGt	MWheel	Vibrato/Tremolo
		Data	Enables Dist Gtr Lyrs
		MIDI22	"Para EQ ^ Hi Freq Stim Drive, Dist EQ"
		MIDI23	"EnvCtl: Imp, Dist Drive"
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) FDR Hall Lvl, Rvb Time
		MIDI26	Flange FB
		MIDI27	Flange Tempo
		MIDI28	Delay Mix
		MIDI29	Delay FB
		MPress	"Vibrato, Harmonics Lvl"
		PWheel	(Dist Lyr) +2/-12 Pitch Bend
138	Liquid T Lead	MWheel	Vibrato
		Data	EnvCtl: Att, LoPass Freq+Res
		MIDI22	"Lopass Freq+Res, Steep Bass Freq"
		MIDI23	EnvCtl: Imp
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Hall Lvl
		MIDI26	"Hall Time+HFDamp, Chorus FB"
		MIDI27	"Delay Mix, SRS EQ"
		MIDI28	"Delay FB, SRS Centerspace"
		MIDI29	toggle: CHDelay + SRS
		MPress	Vibrato, Lyr Enable (Harmonics)
139	Hammeron Synth	MWheel	Steep Resonant Bass Freq
		Data	"Cabinet Preset, Out Gain (KDFX)"
		MIDI22	(Aux) Hall Level
		MIDI23	MD Wet/Dry
		MIDI24	Chorus Wet/Dry
		MIDI25	Bass Tone
		MIDI26	Mid Tone
		MIDI27	Treble Tone
		MIDI28	FX1 Aux Level
		MIDI29	toggle FX
		MPress	"Steep Resonant Bass Freq, Tube Drive"
140	CeeTaur	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI22	EnvCtl: Imp
		MIDI23	EnvCtl: Att
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Hall Lvl, (Fx3) Rvb Time
		MIDI26	(Fx2) Phase W/D
		MIDI27	Phase L/R LFO, (Fx3) Flange Mix
		MIDI28	Delay Mix
		MIDI29	Bus toggle:, Phaser LFO Rate
		MPress	Vibrato
141	Brite Stand-up	MWheel	Vibrato
		Data	"Octave Pitch Shift Layer 2, (Aux) Ambience Level"
142	DualBass^Slp-Bass	MWheel	Vibrato
		Data	toggle: DualBass + SlpBass
		MIDI22	"EnvCtl: Dec, BandPass adj, ParaTreb"
		MIDI23	EnvCtl : Att+Imp
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Room Lvl+Time
		MIDI26	Phaser Notch/ BP ^ Enhc LoDrive+Delay
		MIDI27	Phaser Center Freq L ^ Enhc Hi Mix
		MIDI28	Phaser Center Freq R ^ Enhc Mid Mix
		MIDI29	Phaser FB boost * Enhc Crossover Freq
		MPress	Vibrato
143	Sust Bass	MWheel	Vibrato
		Data	"BandPass Freq+Width, EnvCtl: Imp, LoPass adj"
		MIDI22	EnvCtl: Imp
		MIDI23	EnvCtl: Rel
		MIDI24	In EQ: Bass
		MIDI25	Comp Att Time
		MIDI26	Comp Rel Time
		MIDI27	Comp Ratio
		MIDI28	Comp ThReshhold
		MIDI29	"toggle: Comp I/O, (aux) Room I/O"
		MPress	Vibrato
144	Fonkin Bass	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI24	Bass EQ Gain (KDFX)
		MIDI25	Comp Att Time
		MIDI26	Comp Rel Time
		MIDI27	Comp Ratio
		MIDI28	Comp ThReshhold
		MIDI29	(Aux) Room Level

id	name	ctrl	function
145	Synth Fretless	MWheel	Vibrato
		Data	"Shaper amt, HiPass Freq"
		MIDI22	InEQ: Bass
		MIDI23	EnvCtl: Imp
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Hall Lvl
		MIDI26	"Flange W/D, Chorus W/D"
		MIDI27	"Flange FB, Chorus FB"
		MIDI28	"Flange L/R Phase, Chorus Rate"
		MIDI29	toggle: Flange + Chorus
		MPress	"Vibrato, Shaper adj, Flange W/D"
146	SquashStudio Kit	MWheel	AltControl: Toms
		Data	"Pitch: Kicks, Snares, Toms, HiHats"
		MIDI22	Snare Filters
		MIDI23	Kick Filters
		MIDI24	"EnvCtl: Kicks, Snares, Toms"
		MIDI25	"(FX1+2)- (aux) Room Lvl+Time, (FX2)- Mix Lvl "
		MIDI26	(FX2) Compressor Ratio+Gain
		MIDI27	Room HFDamp
		MIDI28	"toggle: Enhancer HiDrive, Room PreDly"
		MIDI29	Enhancer Hi Delay Time
147	Garage Kit II MW	MWheel	Multiple Layer toggle
		Data	"Pitch: Kicks, Toms"
		MIDI22	"Pitch: Snares, Crash2"
		MIDI23	"EnvCtl: Kicks, Toms"
		MIDI24	"EnvCtl: Snares, HiHats"
		MIDI25	(aux) RoomGate Absorption+Gain
		MIDI26	(FX3) Compression control
		MIDI27	(FX3) InEQ: Treb
		MIDI28	(FX3) InEQ: Bass
		MIDI29	"toggle: (aux) Room type, Lopass adj"
148	Studio Kit II MW	MWheel	Multiple Layer toggle
		Data	"Pitch: Kicks, Toms"
		MIDI22	"Pitch: Snares, Crash2"
		MIDI25	"FX1 Wet/Dry, FX1+2 Aux Levels, Aux Rev Time"
		MIDI26	"FX1 Rev Time,FX2 Wet/Dry"

id	name	ctrl	function
149	General MIDI Kit	MWheel	"Assorted Filters, on most elements"
		Data	"PItch: Kicks (B1, C2), and Toms"
		MIDI22	"Pitch: Snares (D2, E2), HiHats, Ride, Crash (C#3)"
		MIDI23	"Pitch: Congas, Timbales, many other elements"
		MIDI24	EnvCtl / ASR Amp Env: Kicks (above), Snares (above) Toms, Crashes, Ride, Triangle, Ding (A#1)
		MIDI25	(FX1) Room W/D
		MIDI26	Room Rvrb Time
		MIDI27	"(aux) Hall Lvl, (FX1) Mix Lvl"
		MIDI28	(FX1) Compressor Ratio+Threshold+Rel Time
		MIDI29	"toggle: (FX1) Room+Booth, (aux) Hall+""Slither Booth"""
150	ElectroDrum-setGM	MWheel	(FX2) Resonant Filter Freq
		Data	"Filter: Kicks, Toms, assorted other elements"
		MIDI22	"Pitch: Snares, some Toms, Cymbals,+other elements"
		MIDI23	"Filter: Snares, Cymbals, HiHats, Synth Boing"
		MIDI24	EnvCtl: most elements
		MIDI25	"(FX1) Room W/D, (FX3) Echo W/D, (aux) Hall W/D+Lvl"
		MIDI26	"Room Time, (aux) Hall Lvl"
		MIDI27	Hall Late Rvrb Time
		MIDI28	(FX3) Delay Feedback (only a few elements)
		MIDI29	"toggle: Room + ResFilt, Delay + Room"
151	QuestHipKit	MWheel	EP Chord Tremolo
		Data	Low Pass Freq Snare
		MIDI22	EP Chord Low Pass Freq
		MIDI23	EP Chord Resonance
		MIDI25	"FX1,2,3 Aux Level, FX3 Reverb Mix"
		MIDI26	Aux Rev Time
		MIDI27	FX3 Flanger Feedback Level
		MIDI29	"Toggle FX3, FX2"

id	name	ctrl	function
152	e Drums	MWheel	none
		Data	"Pitch Toms, Kicks"
		MIDI22	Pitch Snares
		MIDI23	Para EQ Toms
		MIDI25	(aux) Hall Level
		MIDI26	"FX1 Wet/Dry, Rev Time"
		MIDI27	(aux) Reverb Time
153	SmallKit+Perc MW	MWheel	Cowbell + Shaker Enable
		Data	Pitch: Kit elements (Kick, Snare, HiHats, Toms, Cymbals)
		MIDI22	Pitch: Congas, Timbales, Agogo, Clave, Cowbell (MW)
		MIDI23	Filters: Cabasas, Tambourines, Clave, Agogo, Timbales, Kick, Snare, HiHats, Toms, Cowbell (MW)
		MIDI24	Pitch+Filter: Cabasas, Shaker (MW), Tambourine (F#3, F#4)
		MIDI25	(FX1+2) Rooms W/D
		MIDI26	Rooms' Times
		MIDI27	"(aux) Plate Lvl, (FX4) Mix Lvl, (FX3) Hall W/D"
		MIDI28	Plate Time
		MIDI29	toggle: Room + Hall
154	Steel Drumz	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI 22	Resonance
		MIDI25	FX1 Wet/Dry
		MIDI26	FX2 Wet/Dry
		MIDI27	FX2 Flanger Feedback Leve
		MIDI28	FX2 LFO Tempo
		MIDI29	toggle: to Flanger
155	Trumpet Flourish	Wheel	Shaper
		Data	Low Pass Freq
		MIDI25	"FX1, Aux Reverb Time"
		MIDI26	Aux HF Damping
156	Mr. Parker	MWheel	Vibrato
		Data	LoPass Freq
		MIDI22	LoPass Res
		MIDI23	LoPass Freq
		MIDI24	EnvCtl: Att+Rel
		MIDI25	(aux) Plate W/D
		MIDI26	Plate Time
		MIDI27	Chorus Mix
		MIDI28	Delay Mix (sys)
		MIDI29	"Plate LFO adj, Delay FB"
		MPress	Pitch Bend down
157	Almost Muted (MW)	MWheel	"Vibrato, mute adj"
		Data	LoPass Freq
		MIDI22	HiPass Freq
		MIDI23	EnvCtl: Imp
		MIDI24	EnvCtl: Rel
		MIDI25	(fx1) Room W/D+Time
		MIDI26	Room HFDamp
		MIDI27	InEQ: Bass
		MIDI28	InEQ: Treb
		MIDI29	EQMorph I/O
		MPress	Vibrato
158	Solo Trombone	MWheel	"Vibrato, Low Pass Freq"
		Data	Low Pass Freq
		MIDI23	Env Ctl: Release
		MIDI24	Treble Shelf EQ
		MIDI25	FX1 Wet/Dry, (aux) Hall Level
		MIDI26	FX1 HF Damping
		Mpress	"Vibrato, volume"
159	Trumpets	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI22	InEQ:Bass
		MIDI23	EnvCtl: Imp
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Chamber Lvl
		MIDI26	Chamber Time+HFDamp
		MIDI27	InEQ: Treb
		MIDI28	Chorus FB
		MIDI29	Chorus I/O
		MPress	Swell
160	Flute^WWind Sect	MWheel	Low Pass Freq - Flute
		Data	toggle: Flute^WWind Sect
		MIDI22	Fade in Chiff Layer - Flute
		MIDI25	(Aux) Hall Level (less)
		MIDI26	"FX1 Absorption, HF Damping, Wet/Dry"
		MIDI27	FX3 Wet/Dry, Feedback Level
		MIDI28	FX3 Tempo
		MIDI29	toggle 4tap
		MPress	"Vibrato, WWind Sect"
161	Oboe ^ Eng Hrn	MWheel	Low Pass Freq
		Data	toggle: Oboe ^ Eng Hrn
		MIDI25	"FX1, (aux) Reverb Time"
		MIDI26	(aux) HF Damping
		MPress	Vibrato
162	Clarinet	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI25	FX1 Wet/Dry
		MIDI26	"FX1, Aux Reverb Time"
		MPress	Vibrato

id	name	ctrl	function
163	Bassoon	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI25	FX1 Wet/Dry
		MIDI26	"FX1, Aux Reverb Time"
		MPress	Vibrato
164	Accordion	MWheel	Vibrato
		Data	Enable Layer 3&4
		MIDI22	fade out layer 2
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treble
		MIDI25	FX1 Wet/Dry
		MIDI26	"FX1 Reverb Time,FX2 Feedback Level"
		MIDI27	"FX2 Ctr Freq, LFO Dpth"
		MIDI28	(Aux) HF Damping
		MIDI29	toggle: Room^Phaser
		MPress	Vibrato
165	Matrix 12	MWheel	Vibrato
		Data	Low Pass Freq, Env Ctl: Att, Dec
		MIDI22	Octave Shift Saw+
		MIDI23	Env Ctl: Release
		MIDI24	Impact
		MIDI25	(Aux) Plate Level
		MIDI26	Delay Feedback
		MIDI27	Delay HF Damping (FX3)
		MIDI28	(aux) HF Damping
		MIDI29	Switch in Delay
		MPress	Vibrato
166	OB Brass	MWheel	Vibrato
		Data	LoPass Freq
		MIDI22	LoPass Freq
		MIDI23	"EnvCtl: Attack, Release"
		MIDI24	EnvCtl: Impact
		MIDI25	(aux) Plate Lvl+Time
		MIDI26	Enhc Lo Drive+Mix, Chorus W/D
		MIDI27	"Enhc Mid Drive, Mid Mix"
		MIDI28	"Enhc Hi Drive, Hi Mix, InEQ: Treb"
		MIDI29	toggle: Enhancer + Chorus
		MPress	Vibrato
167	PWM Comper	MWheel	Vibrato
		Data	Env Ctl: Attack
		MIDI22	Env Ctl: Release
		MIDI25	FX1b Reverb Wet/Dry
		MIDI26	(Aux) Hall Level
		MIDI27	FX1a Chorus Wet/Dry
		MIDI28	FX1a Chorus Feedback Level
168	Soft Matrix 12	MWheel	Vibrato
		Data	LoPass Freq
		MIDI22	Pitch Shift - Fifths
		MIDI23	EnvCtl: Decay
		MIDI24	EnvCtl: Release
		MIDI25	(aux) Hall Level
		MIDI26	"(aux) Decay Time, Room Size, HF Damp"
		MIDI27	FX3 Delay Wet/Dry
		MIDI28	FX3 Delay Feedback
		MIDI29	Switch in Delay
		MPress	Vibrato
169	Synth Brass	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI23	Env Ctl: Attack
		MIDI24	Env Ctl: Release
		MIDI25	(Aux) Hall Level
		MIDI26	Chorus Mix
		MIDI27	Delay Mix
		MIDI28	Delay Time
		MIDI29	"toggle Chorus, Env Follower"
		MPress	Low Pass Freq Lyr 1
170	Moogy Bass Too	MWheel	Vibrato
		Data	"Low Pass Freq, Impact"
		MIDI22	Resonance
		MIDI24	Env Ctl: Release
		MIDI25	"(aux) Hall Level, Aux Wet/Dry"
		MIDI26	Chorus Mix
		MIDI27	Chorus Feedback
		MIDI28	FX2 Rev HF Damping
		MIDI29	toggle: Chorus^ Enhancer
		MPress	Vibrato
171	Chirp Bass	MWheel	Vibrato
		Data	HiPass Freq
		MIDI22	LoPass Res
		MIDI23	Env Ctl: Impact
		MIDI24	EnvCtl: Att+Rel
		MIDI25	(aux) Room Lvl
		MIDI26	"Flange W/D, Chorus W/D"
		MIDI27	"Flange FB, Chorus FB"
		MIDI28	"Flange LFO Period, Chorus Tap Delay"
		MIDI29	toggle: Flange + Chorus
		PW	Octave Shift

id	name	ctrl	function
172	Pulsepluck	MWheel	Vibrato
		Data	Pulse Width
		MIDI22	Env Ctl: Attack
		MIDI23	Env Ctl: Impact
		MIDI24	Disable Layer 3
		MIDI25	(Aux) Wet/Dry
		MIDI26	(Aux) Reverb Time
		MIDI27	(Aux) HF Damping
		MIDI28	(Aux) Treble Shelf Freq
173	Resoshape	MWheel	Pitch Modulation
		Data	Shaper Layer 1
		MIDI22	Shaper Layer 2
		MIDI23	Bandpass Width
		MIDI24	Global LFO Rate
		MIDI25	FX1 Wet/Dry, (aux) Wet/ Dry
		MIDI26	FX1 Course Xcursion
		MIDI27	FX1 Flange Feedback Level
		MIDI28	FX1 HF Damping
174	Solar Lead	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI22	Resonance
		MIDI25	(Aux) Wet/Dry
		MIDI26	(Aux) Reverb Time
		MIDI27	(Aux) HF Damping
		MPress	Vibrato
175	Flutey Leads	MWheel	Vibrato
		Data	Pitch - Octave Shift
		MIDI22	InEQ: Bass
		MIDI23	InEQ: Treb
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Hall Lvl, (FX3) Hall Mix
		MIDI26	(aux) Hall HFDamp+PreDly
		MIDI27	Chorus Mix
		MIDI28	Chorus Depth
		MIDI29	toggle: CDR + Room
		MPress	Vibrato
176	TM Lead	MWheel	Octave Harmonic Feed-back
		Data	Low Pass Freq
		MIDI22	Resonance
		MIDI24	! Gain
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Wet/Dry
		MIDI27	FX3 Delay Time
		MPress	Vibrato

id	name	ctrl	function
177	Modular Lead	MWheel	Vibrato
		Data	Octave Pitch Shift Layer 1
		MIDI22	Low Pass Freq, fade Layer 1
		MIDI23	EnvCtl: Att
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Level
		MIDI27	Chorus Feedback
		MPress	Vibrato
178	BrassyFluty Lead	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI22	Resonance
		MIDI24	Env Ctl: Release
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Decay Time, HF Damping
		MPress	Swell
179	Retrosiren	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI22	Env Ctl: Attack
		MIDI23	Env Ctl: Impact
		MIDI24	Env Ctl: Release
		MIDI25	FX1 Wet/Dry
		MIDI26	(Aux) Hall Level
		MIDI27	(Aux) Reverb Time
		MPress	Vibrato
180	Odysseus	MWheel	Vibrato
		Data	Bandpass Freq
		MIDI22	Low Pass Freq
		MIDI23	Sine + Freq
		MIDI24	Low Pass Freq
		MIDI25	FX1 (aux) Wet/Dry
		MIDI26	FX1 Loop Level
		MIDI27	(Aux) Spacing
		MIDI28	(Aux) HF Damping
181	Synth Caliopies	MWheel	Vibrato
		Data	Lyr disable(up); LoPass Res
		MIDI22	BandPass Freq; LoPass Freq
		MIDI23	"LoPass Freq+Res, Hipass Freq, Treb boost"
		MIDI24	EnvCtl: Att+Rel
		MIDI25	"(aux) Hall Lvl, Room W/ D"
		MIDI26	Phaser FB
		MIDI27	Phaser LFO Rate
		MIDI29	"toggle: Room+Phaser(Lyr 1+3), Phaser+CDR(Lyr 2+4)"
		MPress	Vibrato

id	name	ctrl	function
182	Harmonica	MWheel	Vibrato
		Data	InEQ: Bass
		MIDI22	InEQ: Treb
		MIDI23	Env Ctl: Attack
		MIDI24	Env Ctl: Release
		MIDI25	(Fx1) Room W/D, (aux) Hall Lvl
		MIDI26	"Room Time, Phase FB"
		MIDI27	"(aux) Hall adj, Phase Center Freq+LFODepth"
		MIDI28	(aux) Hall HFDamp
		MIDI29	toggle: Room + Phaser
		MPress	Vibrato
183	Space Log	MWheel	Vibrato
		Data	Pitch Shift
		MIDI25	(Aux) Wet/Dry
		MIDI26	(Aux) Rev Time
		Mpress	Vibrato
184	Brite Bells	MWheel	Vibrato
		Data	Gain
		MIDI22	Sine+ Pitch
		MIDI23	Env Ctl: Decay
		MIDI25	FX1 Wet/Dry, (Aux) Room Level
		MIDI29	toggle Hall^Chorus
		Mpress	Vibrato
185	Glasswaves	MWheel	Vibrato
		Data	Non Lin Gain, Low Pass Freq
		MIDI22	InEQ: Bass
		MIDI25	FX1 Wet/Dry, (Aux) Hall Level
		MIDI29	toggle Hall^Chorus
		Mpress	Vibrato
186	Meditator	MWheel	Vibrato
		Data	Low Pass Freq + Res
		MIDI22	"HFstim adj, Lyr Pitch adj"
		MIDI23	Bandpass Freq
		MIDI25	(aux) Hall Lvl + Decay Time ^ Miniverb Lvl
		MIDI26	Flang W/D ^ Minivrb Time + PreDly
		MIDI27	Flange FB
		MIDI28	Delay FB
		MIDI29	toggle: Flange + CDR
		MPress	Vibrato
187	Chariots	MWheel	Vibrato
		Data	Octave Pitch Shift
		MIDI23	Env Ctl: Attack
		MIDI24	Env Ctl: Release
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Reverb Time
		MIDI29	(Aux) Hall Level

id	name	ctrl	function
188	Heaven Stack	MWheel	Vibrato
		Data	Hi Freq Stimulator Drive (less)
		MIDI22	Fade Out Layer 1
		MIDI24	Env Ctl: Release
		MIDI25	(Aux) Room Level
		MIDI26	"(Aux) Reverb Time, FX2 Chorus Wet/Dry"
		MIDI27	FX2 Chorus LFO Rate
		MIDI28	FX2 Chorus Feedback Level
		MIDI29	Switch in FX2 Chorus
		MPress	Vibrato
189	Vortex Rev	MWheel	Vibrato
		Data	HiPass Freqs+Width
		MIDI22	Lyr Xfade
		MIDI23	InEQ: Bass
		MIDI24	"InEQ: Treb, EnvCtl: Att+Rel"
		MIDI25	(aux) Hall Time
		MIDI26	Hall PreDly
		MIDI27	Chorus Depth+Delay
		MIDI28	Delay Mix+FB
		MIDI29	Hall HFDamp
		MPress	Vibrato
190	Luscious	MWheel	Vibrato
		Data	"Panner LFO Rate, Lyr Delay, Lyr Xfade"
		MIDI22	EnvCtl: Imp+Att
		MIDI23	"InEQ: Bass, EnvCtl: Dec"
		MIDI24	"InEQ: Treb, EnvCtl: Rel"
		MIDI25	(aux) Hall Time+PreDly+HFDamp
		MIDI26	Flange Mix
		MIDI27	Flange Rate
		MIDI28	Flange FB
		MIDI29	Hall PreDly adj
		MPress	Vibrato
		Tempo	LoPass Freq
191	Sphaerique	MWheel	Vibrato
		Data	High Pass Freq
		MIDI22	All Pass Freq
		MIDI23	Octave Shift Layer 2
		MIDI24	Env Ctl: Attack
		MIDI25	(Aux) Wet/Dry
		MIDI26	(Aux) Reverb Time
		MIDI27	(Aux) HF Damping
		MIDI28	InA Bass EQ

id	name	ctrl	function
192	Padifier	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI22	Env Ctl: Attack
		MIDI23	Env Ctl: Release
		MIDI24	FX3 Delay Mix
		MIDI25	"(Aux) Hall Level, FX3 Reverb Mix"
		MIDI26	"(Aux), FX3 Reverb Time"
		MIDI27	FX3 Chorus Mix
		MIDI28	FX3 Chorus Depth
		MIDI29	toggle: Chorus^Room
		Mpress	Vibrato
193	Tang Vox Pad	MWheel	Vibrato
		Data	Pitch Layer 2
		MIDI22	Low Pass Freq
		MIDI23	"Env Ctl: Att, Rel"
		MIDI24	Env Ct: Decay
		MIDI25	(Aux) Wet/Dry
		MIDI26	(Aux) Reverb Time
		MIDI27	(Aux) HF Damping
		MIDI28	(Aux) Treble Shelf Freq
		Mpress	Vibrato
194	Interference	MWheel	Wrap
		Data	Shaper
		MIDI25	(Aux) Room Level
		MIDI26	(Aux) HF Damping
		MIDI27	FX1 4 Tap Mix
195	One Shot	MWheel	Vibrato
		Data	LPGate Freq
		MIDI22	Saw+ Pitch Layer 1
		MIDI23	Saw+ Pitch Layer 2
		MIDI24	Env Ctl: Release
		MIDI25	"FX3, Aux Wet/Dry (dryer)"
		MIDI26	(Aux) Hall Level (less)
		MIDI27	FX3 Tap Delays, Loop Length
		MIDI28	FX3 HF Damping
196	Integrated Circuit	MWheel	Saw+ Pitch
		Data	Hi Pass Freq
		MIDI22	Saw+ Pitch
		MIDI23	Low Pass Freq
		MIDI24	Env Ctl: Attack
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Rev Time
		MIDI27	Chorus Feedback
		MIDI28	Chorus Depth
197	Doomsday	MWheel	"Pitch, Shaper Layer 2"
		MIDI25	(Aux) Hall Level
		MIDI26	(Aux) Decay Time
198	Click		
199	Default Program		

id	name	ctrl	function
730	L'il Nipper Kit	MWheel	SFX Pitch
		Data	"Pitch: Kick, Toms"
		MIDI22	"Pitch: Snares, AuxPerc"
		MIDI23	"Filter: Hihats, Cymbals"
		MIDI24	"EnvCtl: Kicks, Snares"
		MIDI25	(aux) Plate Time
		MIDI26	(FX3) Laserverb Spacing
		MIDI27	(FX2) Pitcher Pitch, Pitcher W/D
		MIDI28	Pitcher control
		MIDI29	Laserverb Delay+Contour+FB
		MPress	AuxPerc Pitch
731	Industry Set II	MWheel	Filter Resonance (A#4-C5)
		Data	"AltControl on some layers," Pitch on Kick-like elements and some Toms
		MIDI22	Various Pitch controls on many elements
		MIDI23	Filters or Modulation Pitch on many elements
		MIDI24	EnvCtl: assorted kinds of control on many elements
		MIDI25	(FX2) Flange W/D, InEQ: Bass
		MIDI26	"(aux) Hall Lvl, (FX2) Mix Lvl"
		MIDI27	(FX3) DistEQ W/D+Gain Adjust
		MIDI28	Distortion Warmth
		MIDI29	toggle: RoomType: Hall + Delay
		MPress	Filter Resonance (A#4-C5)
732	Technoo Kit	MWheel	Alternate Kick (B2-C3)
		Data	Pitch: nearly all elements
		MIDI22	"Filter: Kicks, AuxPerc"
		MIDI23	"Filter: Snares, Toms, Ride, Crashes, HiHats (A#1-B1)"
		MIDI24	"EnvCtl: Kicks, Snares (not G#1-A1), Ride, Choke Cym"
		MIDI25	(FX1) Gated Reverb W/D
		MIDI26	Gated Reverb Time
		MIDI27	(FX1+2) (aux) LaserVerb Lvl
		MIDI28	(FX4) LaserVerb Lvl
		MIDI29	toggle: GateRvb HFDamp+Gate Threshold

id	name	ctrl	function
733	Geo-Kit MW+22	MWheel	Multiple Layer toggle
		Data	"Pitch: Kicks, Snares, Toms, ""Shaker"""
		MIDI22	Crossfade to tertiary Kicks; Pitch: Elec. Snare only
		MIDI23	Filter: Kicks, Snares, HiHats, Crashes, Ride, Shaker Amp LFO: SFX (A6-B6)
		MIDI24	EnvCtl: most Kicks, Snares, Toms, Shaker, Elec HiHat LFO Rate: SFX (A6-B6)
		MIDI25	(FX3) Mix Lvl, (aux) GateRvb Lvl
		MIDI26	"(FX4) Mix Lvl, GateRvb Lvl"
		MIDI27	(FX3) Compressor Smooth-Time+MakeUpGain
		MIDI28	(FX2) EnvFlt Freq Sweep+Threshold, (FX1) Delay Lvl
		MIDI29	toggle: Compressor + ChorDelay
734	Slam 'n Drums I	MIDI25	"(FX1) Rev Time, Wet/Dry, HF Damping"
		MIDI26	"(FX1) Aux Level, InA EQ Treb"
		MIDI27	"(FX2, FX4) Aux Level"
		MIDI28	(FX2) Wet/Dry
735	Bottom-Feed^Pulse	MWheel	Vibrato
		Data	toggle: BottomFeed ^ Pulse
		MIDI22	"LoPass Gate+Freq, EnvCtl: Imp+Att"
		MIDI23	EnvCtl: Att+Dec, Saw Pitch adj
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Room Lvl, (FX3)Hall Mix
		MIDI26	Chorus Mix
		MIDI27	Chorus Rate
		MIDI28	Chorus FB
		MIDI29	toggle: Chorus(4Tap) + Flange
		MPress	Vibrato
736	Lowdown Bass	MWheel	"Vibrato, HiPass Freq (Chirp)"
		Data	LoPass Gate
		MIDI22	EnvCtl: Imp
		MIDI23	EnvCtl: Att
		MIDI24	"Lyr Enable, EnvCtl: Dec+Rel"
		MIDI25	(aux) Dist Lvl
		MIDI26	"Dist Drive, Mid EQ cut, Flange W/D"
		MIDI27	"InEQ: Bass, Flange FB"
		MIDI28	Cab HiPass
		MIDI29	toggle: EQ + Flange
		MPress	Vibrato
737	SustBass^Mix-Bass	MWheel	"Vibrato, LoPass Freq"
		Data	toggle: SustBass + MixBass
		MIDI22	"BandPass Freq+Width, EnvCtl: Imp, LoPass adj"
		MIDI23	EnvCtl: Rel
		MIDI24	In EQ: Bass
		MIDI25	Comp Att Time
		MIDI26	Comp Rel Time
		MIDI27	Comp Ratio
		MIDI28	Comp Threshhold
		MIDI29	"toggle: Comp I/O, (aux) Room I/O"
		MPress	Vibrato
738	SkoolBass^SIm-ple	MWheel	Vibrato
		Data	toggle: SkoolBass ^ SImple
		MIDI22	"Pulse Width+Freq, Pitch adj, EnvCtl: Imp+Att"
		MIDI23	"Dist Drive adj, EnvCtl: Dec"
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Room Lvl
		MIDI26	Phase Notch/ Dry, Dist W/ D"
		MIDI27	"Phase Center Freq, Dist Drive adj
		MIDI28	Phase LFO Depth, Dist Bass adj
		MIDI29	"toggle: Phase + Dist, Room Time adj"
		MPress	Vibrato
		AttVel	LoPass gate
		GKey-Num	L/R Phase
739	Default Triple		

id	name	ctrl	function
770	Mellostr^ShineOn	MWheel	Vibrato
		Data	toggle: Mellostr ^ ShineOn
		MIDI22	LoPass+BandPass Freq+Width
		MIDI23	"EnvCtl: Att, LoPass Res"
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Room Lvl, Hall absorption
		MIDI26	"Filt Res, Chorus FB"
		MIDI27	"Filt Freq, Chorus Rate"
		MIDI28	"Filt Vibrato, Delay Mix"
		MIDI29	toggle: Res Filt + ChorDelay (Mellostr only)
		MPress	"Vibrato, HiPass Freq"
771	Arystal^InTheAir	MWheel	Vibrato
		Data	toggle: Arystal ^ InTheAir
		MIDI22	Lyr Pitch adj ^ LoPass adj
		MIDI23	"LoPass Freq ^ Saw Pitch, Lyr detune"
		MIDI24	"Lyr Pitch adj, Lyr Xfade"
		MIDI25	(aux) Hall Lvl+Time
		MIDI26	Chorus W/D
		MIDI27	Chorus FB
		MIDI28	Chorus Rate
		MIDI29	"ChorusDelay I/O (sys), InEQ: Treb boost"
		MPress	Vibrato
		ControlD	amp cut
772	Padify	MWheel	Vibrato
		Data	none
		MIDI22	LoPass Freq
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treb
		MIDI25	(aux) Hall Lvl
		MIDI26	Chorus Delay Time
		MIDI27	Chorus Delay Depth
		MIDI28	Delay Mix (sys)
		MIDI29	Hall Time+PreDly adj
		MPress	Vibrato
773	OronicoKno^Shift	MWheel	Vibrato
		Data	toggle: OronicoKno + Shift
		MIDI22	"HFstim adj, Pan adj"
		MIDI23	"InEQ: Bass, Lyr Xfade"
		MIDI24	InEQ: Treb, Pan adj, EnvCtl: Rel
		MIDI25	(aux) Hall Lvl
		MIDI26	Hall Decay Time+PreDly
		MIDI27	Delay Mix (sys)
		MIDI28	Chorus Delay Time
		MIDI29	Chorus Depth adj
		MPress	Vibrato
		AttVel	AltCtl
774	WispSingrs^Glass	MWheel	"Vibrato, LoPass Res"
		Data	toggle: WispSingrs + Glass
		MIDI22	LoPass Freq+Res, HiPass Freq
		MIDI23	"LoPass Freq, HiPass Res+Freq, Lyr Lvls"
		MIDI24	EnvCtl: Att+Rel
		MIDI25	(aux) Hall + (fx1) Hall W/D
		MIDI26	Hall Times+HFDamp
		MIDI27	Chorus W/D
		MIDI28	Delay W/D (sys)
		MIDI29	toggle: Hall + CDR
		MPress	Vibrato
775	Cymbal Singers	MWheel	Vibrato
		Data	Lyr 3 volume (ride cymbal)
		MIDI22	"BandPass Width, HiPass Res"
		MIDI23	Pan LFO adj
		MIDI24	InEQ: Treb cut
		MIDI25	(aux) LaserVrb Lvl
		MIDI26	LaserVrb contour
		MIDI27	Pitch LFO Rate
		MIDI28	Flange FB
		MIDI29	toggle: Pitcher + Pitcher-Flange
		MPress	"Vibrato, BandPass Freq"
		KeyNum	EnvCtl: Att+Dec
		GKeyNum	Pitcher Pitch+Weights
		PWheel	BandPass Freq
776	Mad Three-O	MWheel	Vibrato
		Data	Low Pass Freq
		MIDI22	"Resonance, 4Pole LP Separation, Distortion"
		MIDI23	Low Pass Freq
		MIDI24	Env Ctl: Decay
		MIDI25	Xfade
		MIDI27	(Aux) Level
		MIDI28	FX3 Delay Mix
		MIDI29	"FX3 Flange Mix, Rvb Mix"
		MPress	Vibrato

id	name	ctrl	function
777	AlaskaGlide (MW)	MWheel	toggle: Alaska + Glide
		Data	EnvCtl: Imp
		MIDI22	EnvCtl: Att
		MIDI23	EnvCtl: Dec
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Hall Lvls
		MIDI26	FDR W/D
		MIDI27	InEQ: Bass
		MIDI28	InEQ: Treb
		MIDI29	FlgDelayrvb I/O
		MPress	"Vibrato, Lyr detune, LoPass Freq, Flange XCurs + FB"
778	Detooner^BigP MW	MWheel	Vibrato
		Data	toggle: Detooner ^ BigPMW
		MIDI22	"P5th jump ^ LoPass Freq, EnvCtl: Att+Rel"
		MIDI23	"Notch Freq ^ Dist drv, EnvCtl: Imp"
		MIDI24	"PWM Width, Dist drv"
		MIDI25	(aux) Laser Lvl
		MIDI26	(aux) Laser contour+FB
		MIDI27	"Flange FB+L/R phase, Phaser Ctr Freq"
		MIDI28	Flange W/D cut, Phaser W/D
		MIDI29	toggle: Flange + Phaser
		MPress	Vibrato
779	Razor Saw	MWheel	Vibrato
		Data	"LoPass LFO Rate, Shaper amt, EnvCtl: Att+Dec "
		MIDI22	EnvCtl: Rel
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treb
		MIDI25	(aux) Hall Lvl+PreDly+Time+HFDamp
		MIDI26	Delay FB+Mix
		MIDI27	Chorus Depth+Rate
		MIDI28	Chorus FB
		MIDI29	toggle: Delay I/O
		MPress	Vibrato

id	name	ctrl	function
780	DynOrch^WTellOrc	MWheel	string and brass balance
		Data	toggle: DynOrch ^ WTellOrch
		MIDI22	"ParaMid and LoPass Freq, Shaper Drive"
		MIDI23	"Shaper amt, LoPass Freq"
		MIDI25	(aux) Hall Lvl cut
		MIDI26	Chapel + Hall Times
		MIDI29	toggle: Chapel/Hall + Hall/Room
		MPress	"(DynOrch) Volume swell, shaper amt"
		SostPd	"Lyr enable, Room Time"
781	OrcBrs^French-Bone	MWheel	Vibrato
		Data	toggle: OrcBrs ^ French-Bone
		MIDI22	InEQ: Bass
		MIDI23	"InEQ: Treb, LoPass Freq"
		MIDI24	EnvCtl: Imp + Rel
		MIDI25	(aux) Hall Mix
		MIDI26	"Hall Time, Mix adj, Pan Rate(Fx3)"
		MIDI27	Chorus Mix
		MIDI28	Delay Mix
		MIDI29	"Hall PreDly, Pan I/O"
		MPress	"Swell, Vibrato Depth"
782	Synth Bell 1^2	MWheel	"Vibrato, Pan adj, LoPass Res"
		Data	"toggle: Synth Bells 1 + 2, AltCtl adj"
		MIDI22	"LoPass Res, BandPass Width, EnvCtl: Rel"
		MIDI23	Pan adj
		MIDI24	Pitch LFO adj
		MIDI25	"(aux) Hall Lvl, (fx1) Chapel W/D"
		MIDI26	"Hall HFDamp+Time, Chapel Time"
		MIDI27	"Chapel preDelay, SRS center Freq adj"
		MIDI28	"Chapel EarlyRef+Late Lvls, SRS EQ adj"
		MIDI29	toggle: Chapel + SRS
		BKeyNu	EnvCtl: Att+Dec+Rel
		MPress	Vibrato

id	name	ctrl	function
783	Crystaline^RX7	MWheel	"Shaper ctl, Vibrato ^ Pan adj"
		Data	toggle: Crystaline ^ RX7
		MIDI22	"ShapeMod osc Pitch, Shape amt ^ LoPass Freq, Pitch adj"
		MIDI23	"LoPass Res, EnvCtl: Att"
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Room Lvl
		MIDI26	Room Decay Time+HFDamp
		MIDI27	"Chorus W/D, Echo W/D"
		MIDI28	"Chorus FB, Echo FB"
		MIDI29	toggle: Chorus + Echo
		MIDI70	Lyr AltCtl
784	Enter-prize^MTree	MWheel	"Vibrato, Tremolo"
		Data	toggle: Enterprize ^ MTree
		MIDI22	"Pitch jump, HFStim ^ EnvCtl: Att+Dec"
		MIDI23	HiPass Freq, Dist Drive (VAST)
		MIDI24	DSP XFade, Pitch adj, EnvCtl: Rel
		MIDI25	(aux) Acid Room Lvl
		MIDI26	"Acid dry Lvl cut, Dist Drive adj ^ LasrVrb W/D"
		MIDI27	Dist warmth ^ LasrVrb Delay Time
		MIDI28	Dist Freq adj ^ LasrVrb contour
		MIDI29	Distortion I/O
		MPress	"Vibrato, Tremolo"
		AttVel	EnvCtl: Rel
785	RaveStrg^Solina	MWheel	"Vibrato, Lyr detune(Sol)"
		Data	toggle: RaveStrg ^ Solina
		MIDI22	EnvCtl: Att+Rel
		MIDI23	"EnvCtl: Dec ^ Ptch mod, Notch LFO Rate"
		MIDI24	"Flange Mix, Spin W/D"
		MIDI25	(aux) Room Lvl
		MIDI26	Spin Pitcher Mix ^ MovDelay W/D
		MIDI27	Spin Pitcher Weights
		MIDI28	Spin Pitcher ptch (rapid echo Rate)
		MIDI29	"toggle: Spin I/O, Room HFDamp+Time"
		MPress	Vibrato
786	Mellotron (MW)	MWheel	"3-way toggle: Ens Strg, Solo Strg(dwn 8ve), Flute"
		Data	Octave jump
		MIDI22	LoPass Freq; ParaTreb Freq ; HiFreqStim Freq
		MIDI23	"Dist Drv, Xfade dpth; ParaTreb dpth; HFStim Drv"
		MIDI25	(aux) Hall Lvl
		MIDI26	Hall Time
		MIDI27	Room Lvl
		MIDI28	Room Time
		MPress	Vibrato
787	Funk O Matic	MWheel	"Vibrato, Vibrato Rate"
		Data	LoPass Freq
		MIDI22	"Shaper amt, LoPass Freq cut"
		MIDI23	Dist Drive
		MIDI24	(Lyr 1+3) 8ve drop
		MIDI25	Env Filt thReshold
		MIDI26	Env Filt min Freq
		MIDI27	(aux) Sweep Filt W/D
		MIDI28	(aux) Sweep Filt min Freq
		MIDI29	toggle: Env Filt - BandPass and HiPass
		MPress	"Vibrato, Lyr detune"
788	Buzz Kill	MWheel	Pitch modulation
		Data	LoPass Freq
		MIDI22	LoPass Res cut, Dist Drive cut
		MIDI23	"EnvCtl: Att, Flange LFO"
		MIDI24	EnvCtl: Rel, Flange L/R phase
		MIDI25	Flange Delay Tempo
		MIDI26	Flange FB
		MIDI27	(aux) CDR Lvl cut
		MIDI28	(aux) Delay Mix
		MIDI29	(aux) Hall W/D+Time adj
		MPress	LoPass Freq
789	Grand+Elec 1	MWheel	Lyr Balance
		MIDI25	"(aux) Hall Lvl, Room W/D"
790	Fluid Grand	Data	Wet/Dry Mix
		MIDI25	(Aux) FDR Level
		MIDI26	(Aux) Wet/Dry, Delay Mix
		MIDI27	(Aux) Flanger Mix
		MIDI28	(Aux) Flanger tempo
791	Haunted Piano	MWheel	Harp Balance
		MIDI25	(Aux) FLRev Level, Rev Time
		MIDI26	Flange Tempo
		MIDI27	(FX3) Wet/Dry

id	name	ctrl	function
792	DrkPno^Arak-isPno	MWheel	Vibrato (ArakisPno)
		Data	toggle: DrkPno ^ Arak-isPno
		MIDI22	detune
		MIDI25	(aux) Chorus/Plate Lvl + W/D
		MIDI26	Plate Time
		MIDI27	Chorus FB
		MIDI28	Chorus Mix
		MPress	Vibrato (ArakisPno)
793	Funky Piano	MWheel	ParaEQ LFO Depth
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treb
		MIDI25	(aux) Room Lvl+Time
		MIDI26	Flange W/D
		MIDI27	Flange FB
		MIDI28	Flange XCouple
		MIDI29	Flange LFO Tempo
		MPress	ParaEQ Depth
794	Water Piano	MWheel	Vibrato
		MIDI25	(Aux) FDR Level
		MIDI26	(Aux) Wet/Dry, Delay Mix
		MIDI27	(Aux) Flanger Mix
		MIDI28	(Aux) Flanger tempo
		Mpress	Vibrato
795	Piano Chase	MWheel	Vibrato (Strings)
		MIDI23	InEQ: Bass
		MIDI24	InEQ: Treb
		MIDI25	(aux) Plate Lvl+Time
		MIDI26	Flange W/D
		MIDI27	Flange FB, aux Decay Time
		MIDI28	Flange LFO Tempo
		MIDI29	Flange XCouple
		MPress	Vibrato (Strings)
		Sost Ped	Disables Strings
796	Noise Toys	MWheel	"Pitch LFO, Shaper amt"
		Data	"Pitch (Sine+) adj, Band-Pass Freq, Dist amt"
		MIDI22	"Pitch adj, Shaper LFO, HiPass Freq"
		MIDI23	"LoPass + HiPass Freq, EnvCtl: Att"
		MIDI24	EnvCtl: Rel
		MIDI25	(aux) Hall Lvl
		MIDI26	"LrsDelay W/D, Pitch W/D"
		MIDI27	"LsrDelay contour, Pitch pair weights"
		MIDI28	Pitch odd weights
		MIDI29	toggle: Laser + Pitch
		MPress	"Vibrato, Pitch LFO adj"
		PWheel	Shaper adj
		Tempo	LsrDelay Delay coarse + spacing

id	name	ctrl	function
797	Environments	MWheel	"hi bird" LFO Rate, Pan adj
		Data	"lo bird" LFO Rate"
		MIDI22	"ParaEQ Freq, shaper amt"
		MIDI23	"Pitch adj, LoPass Freq, BandPass Freq+Width"
		MIDI24	"HiPass Freq, Pitch (sine)"
		MIDI25	Chorus Lvl, rvb Lvl, CDR W/D
		MIDI26	(fx2) Chorus W/D
		MIDI27	Phaser W/D
		MIDI28	CDR W/D
		MIDI29	Chorus Rate
		MPress	InEQ: Bass
		MIDI70	AltCtl
798	Lunar Wind	MWheel	LoPass Freq+Res
		Data	Pitch adj
		MIDI22	"LoPass Res, Pan adj"
		MIDI23	Panner sweep
		MIDI25	(aux) Room Lvl
		MIDI26	Pitcher W/D
		MIDI27	Flange Mix (sys)
		MIDI28	Pitcher Pitch
		MIDI29	toggle: Pitcher I/O
		MPress	"LoPass Freq, Pan LFO"
		ChanS	EnvCtl: Rel
		Breath	LoPass adj
799	Gremlin Groupies	MWheel	"Lyr Pitch, LoPass Freq+Res, Wrap adj"
		Data	"Lyr Pitch, LoPass LFO adj"
		MIDI22	"Lyr Pitch, Pitch (Sine) adj"
		MIDI23	Lyr Pitch adj
		MIDI24	"Lyr Pitch, Wrap adj"
		MIDI25	(aux) Hall Lvl
		MIDI26	"Pitcher W/D, LsrDelay Time+W/D"
		MIDI27	"Pitcher wts pair, Lsr Spac-ing"
		MIDI28	"Pitcher wts odd, Lsr Con-tour"
		MIDI29	toggle: Pitcher + LaserDe-lay

Appendix D
Contemporary ROM Block Objects

This Appendix describes the Contemporary ROM objects provided with your K2661.

Programs

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 K2661 Contemporary ROM 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
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	
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				

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
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)		
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

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
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	
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				

Controller Assignments: Contemporary ROM Block

This supplement lists the controller assignments for all programs and setups in the Contemporary ROM sound block.

Secondary Effects

Some of the programs in the Contemporary block use a programming technique called *secondary effects*, in which the processing on one or more layers of the program can be changed with the press of a button. Secondary effects in these programs are enabled by PSw2 (or by any physical controller assigned to send MIDI 29). PSw2 acts as a toggle between the primary effect and the secondary effect. It switches off one of the two FXBus sends on an Input page (sets its Lvl parameter to **Off**), and simultaneously turns on the other FXBus send (sets its Lvl parameter to **0.0 dB**).

The following diagram shows the effect of pressing PSw2 on the settings for FXBus1 and FXBus2.

PSw2 Status	Value of Lvl Parameter on Input Page	
	FXBus1	FXBus2
Off	0.0 dB	Off
On	Off	O.0 dB

In most cases, toggling effects with PSw2 affects only a single layer on a single input pair. In some cases, however, the switching is more complicated, and toggling effects moves one or more layers to different FX buses. Toggling effects may also change EQ settings, or the Aux reverb's decay time, depending on the program.

The following segment from the controller listings shows an example of secondary effects. Secondary effects appear in italics. In this example, when PSw2 is off, the program's input routings result in a room reverb effect, Slider B controls the wet/dry mix of this reverb. When PSw2 is on, the routing changes, resulting in a flange effect. In this case, Slider B is inactive, Slider C controls the aux room reverb level, and Slider D controls both the flange level and the crosscouple amount.

Program		Studio		Controller Assignments	
ID	Name	ID	Name		
999	SuperSynth	9	RmFlgChDly Room	B	room1 reverb wet/dry
				C	*aux room reverb level*
				D	*flange level, flange Xcouple*
				PSw2	toggle: room1 reverb/*flange*

Program Control Assignments

Program		Studio		Controller Assignments	
ID	Name	ID	Name		
800	Jungle Jam	62	BthQFlg4Tap Hall	B	hall reverb level (FX1+FX2)
				C	hall reverb level (FX4)
				E	*quantization dynamic range*
				F	*flange feedback*
				G	*flange tempo*
				H	*quantization wet/dry*
				PSw2	quantization + flange in/out
801	Mbira Stack	99	auxPhsrFldblHall	B	hall reverb level
				C	hall reverb level
				E	*phaser LFO rate & center frequency*
				F	*phaser rate scale*
				PSw2	phaser *in*/out, EQ treble boost
802	Ritual Metals	39	RmDsRotFl4t RvCm	B	chamber reverb level, *chamber reverb level*
				C	room reverb wet/dry
				D	*chamber reverb level*
				E	*Lo & Hi rate*
				PSw2	toggle: room reverb/*rotary + distortion*
803	Prepared Mbira	7	RoomFlgEcho Hall	B	room reverb wet/dry & time
				C	hall reverb level & time, *flange wet/dry*
				D	*flange feedback level*
				E	*flange LFO tempo*
				F	*hall reverb level & high-frequency damp, flange high-frequency damp*
				PSw2	toggle: room reverb/*flange*
804	Balinesque	7	RoomFlgEcho Hall	B	room reverb wet/dry
				C	hall reverb level (hybrid pan)
				D	echo wet/dry (hybrid pan)
				E	*hall reverb level*
				F	*flange wet/dry*
				G	*flange feedback level*
				H	*flange LFO tempo*
				PSw2	toggle: room reverb/*flange*
805	Ambient Bells	94	auxChorMDly Hall	B	hall reverb level
				C	delay wet/dry
				PSw2	MDly in/*out*, EQ parameters
806	World Jam 1	34	RoomCmpChor Hall	B	room reverb wet/dry
				C	room reverb size scale
				D	*hall reverb level*
				PSw2	toggle: room reverb/*comp*
807	World Jam 2	3	RoomChorCDR Hall	B	hall reverb level
				C	room reverb time
				D	hall reverb decay time
				E	*hall reverb level*
				PSw2	toggle: room reverb/*chorus*
808	India Jam	27	RoomSRSRoom Room	B	aux room reverb level (C0 - F5)
				C	aux room reverb level (F#5 - C 8)
				D	*aux reverb level (C0 - F5)*
				PSw2	toggle: room reverb & *SRS*
809	Slo Wood Flute	69	auxPtchDst+ Chmb	B	chamber reverb level
				C	chamber reverb time
				PSw2	*adds pitcher*
810	Hybrid Pan Flute	7	RoomFlgEcho Hall	B	hall reverb level, *hall reverb level*
				C	room reverb time
				D	room reverb high-frequency damp
				PSw2	toggle: room reverb/*flange*
811	Chiff Brass Lead	26	RoomSrsCDR Hall	B	hall reverb level
				C	room reverb wet/dry, reverb time (synth brass)
				D	*delay level*
				PSw2	toggle: SRS/*CDR* (pan flute)

Program ID	Program Name	Studio ID	Studio Name	Controller	Assignment
812	Bell Players	11	RoomFlngCDR Hall	B	hall reverb level
				C	room reverb & flange wet/dry
				PSw2	toggle: room + flange/*flange + CDR*
813	Prs Koto	9	RmFlgChDly Room	B	room1 reverb wet/dry
				C	*aux room reverb level*
				D	*flange level, flange Xcouple*
				PSw2	toggle: room1 reverb/*flange*
814	Medicine Man	7	RoomFlgEcho Hall	B	hall reverb level, room reverb cut
				D	*hall reverb level*
				E	*flange LFO tempo*
				PSw2	toggle: room reverb/*flange*
815	Mbira	7	RoomFlgEcho Hall	B	room reverb wet/dry
				D	*hall reverb level*
				F	*flange feedback level*
				G	*flange LFO tempo*
				PSw2	toggle: room reverb/*flange*
816	Kotobira	11	RoomFlngCDR Hall	B	hall reverb level
				D	*hall reverb level*
				E	*flange feedback level*
				F	*flange LFO tempo*
				G	*flange Xcouple*
				PSw2	toggle: room reverb/*flange*
817	Cartoon Perc	62	BthQFlg4Tap Hall	B	booth reverb wet/dry
				D	*hall reverb level*
				E	*quantization + flange level (dynamic range)*
				PSw2	toggle: booth reverb/*quantization + flange*
818	CowGogiBell	76	HallGateFl4T Bth	B	booth reverb level
				C	hall reverb wet/dry
				D	booth reverb time
				E	*booth reverb level*
				PSw2	toggle: hall/*gate*
819	Perc Pan Lead	98	auxFlngCDR Hall	B	hall reverb level & time
				C	delay mix
				D	*hall reverb level*
				PSw2	CDR in/*out*, EQ treble boost
820	Trippy Organ	126	GtRvShapMDl Room	B	gated reverb gate time
				C	gated reverb reverb time
				D	*shaper amount*
				PSw2	toggle: gated reverb/*shaper*
821	Koto Followers	3	RoomChorCDR Hall	B	hall reverb level
				C	*CDR reverb mix, hall reverb level*
				D	*delay mix*
				E	*delay feedback*
				F	*chorus feedback*
				PSw2	toggle: chorus/*CDR*
822	Hybrid Horn	10	ChmbFlgGtRv Hall	B	hall reverb level
				C	flange wet/dry
				D	*hall reverb level*
				E	*gated reverb wet/dry*
				F	*gate time*
				G	*gate release time*
				PSw2	toggle: flanger/*gated reverb*
823	Dyno EP Lead	3	RoomChorCDR Hall	B	CDR reverb time
				C	CDR delay mix
				D	hall reverb level
				E	*hall reverb level*
				F	*hall reverb wet/dry, time & high-frequency damp*
				PSw2	toggle: CDR/*room reverb*
824	ParaKoto	92	auxFlgDist+ Hall	B	hall reverb level
				C	flange wet/dry
				D	*hall reverb level*
				PSw2	toggle: flange/*distortion*

Controller Assignments: Contemporary ROM Block

Program		Studio		Controller Assignments	
ID	Name	ID	Name		
825	Super Clav	92	auxFlgDist+ Hall	B	hall reverb level
				C	flange feedback level
				D	*delay wet/dry*
				PSw2	toggle: flange/*distortion+delay+chorus*
826	StrataClav	92	auxFlgDist+ Hall	B	hall reverb level
				C	flange feedback level
				PSw2	toggle: flange/*distortion+delay+chorus*
827	Touch Clav	92	auxFlgDist+ Hall	B	hall reverb level
				C	flange wet/dry & feedback level
				PSw2	toggle: flange/*distortion+delay+chorus*
828	Bad Klav	91	auxChrDist+ Hall	B	hall reverb level
				C	*chorus feedback level*
				D	*reverb level*
				PSw2	chorus *in*/out
829 830 831 832	Rad Rotor B-2001 Perc Organ Drawbar Organ CS	145	auxRotaryFDR Plt	B	vib+chorus in/out, vib/chorus config
				C	plate reverb level
				D	plate reverb time
				E	rotary hi & lo gain
				F	rotary trem level
				G	plate reverb high-frequency damp
				MWheel	rotary rate
				PSw2	toggle: rotary/*FDR*
833	Bebop Alto Sax	25	RmRotoFl4T CmpRv	B	room reverb wet/dry, reverb time
				C	aux comp & reverb level
				MW	rotor speed
				PSw2	toggle: room reverb/*rotary effect*
834	Soft Alto Sax	65	ChamDstEcho Room	B	room reverb level
				C	room reverb time
				D	chamber wet/dry
				E	*room reverb level*
				F	EQ treble boost
				PSw2	toggle: chamber & *distortion, EQ*
835	Soprano Sax	63	ChmbTremCDR Room	B	CDR reverb level
				C	CDR chorus mix
				D	CDR delay mix
				E	*room reverb level*
				F	*chamber reverb level*
				G	EQ treble cut
				PSw2	toggle: CDR/*chamber reverb*
836	Low Soft Sax	6	RoomFlngCDR Hall	B	hall reverb level
				C	room reverb wet/dry
				D	room reverb time
				E	EQ treble boost
				F	*hall reverb level*
				G	*flange wet/dry*
				H	*flange feedback level*
				PSw2	toggle: room reverb/*flange*
837	Air Reeds CS	34	RoomCmpChor Hall	B	room reverb wet/dry & hall reverb level
				C	room reverb time
				D	room reverb high-frequency damp
				E	*hall reverb level*
				PSw2	toggle: room reverb & *compressor*
838	Jazz Muted Trp	23	RmSweepEcho Hall	B	room reverb wet/dry, hall reverb level, *hall reverb time*
				C	room reverb time
				D	room & hall reverbs high-frequency damp
				E	*hall reverb level*
				PSw2	toggle: room reverb/*LFO filt sweep*

Program ID	Program Name	Studio ID	Studio Name	Controller	Assignments
839	Jazz Lab Band	3	RoomChorCDR Hall	B	room reverb wet/dry, hall reverb level
				C	room reverb time
				D	room reverb high-frequency damp
				E	*hall reverb level*
				F	*chorus wet/dry*
				G	*chorus feedback level*
				PSw2	toggle: room reverb/*chorus*
840	Harmon Section	73	auxChorFlRv Cmb4	B	chamber reverb level
				C	chamber reverb absorption, high-frequency damp, treble cut
				D	*chamber reverb level*
				E	*chorus feedback level*
				F	*chorus wet/dry*
				PSw2	chorus *in*/out
841	Sfz Cres Brass	111	PltEnvFl4T Room	B	plate reverb wet/dry, room reverb level
				C	room reverb high-frequency damp, lopass frequency
				D	*room reverb level*
				F	*env filt resonance*
				G	*env filt minimum frequency*
				GAttVel	*env filt frequency sweep range*
				PSw2	toggle: plate reverb/*env filt*
842	Neo Stabs	127	GtdEnhcStIm Room	B	room reverb level
				C	room reverb time
				D	gate reverb wet/dry, room reverb pre-delay
				E	gated reverb gate release rate
				F	*room reverb level*
				G	*enhancer EQ high boost*
				PSw2	toggle: gated reverb/*enhancer*
843	Gtr Jazz Band	42	RoomRmHall Hall	B	hall reverb level
				C	room1 reverb wet/dry (bass & drums)
				D	room2 reverb wet/dry (gtr & horns)
				E	room2 reverb time (gtr & horns)
				PSw2	*room2 size (gtr & horns)*
844	Full Rock Band	25	RmRotoFl4T CmpRv	B	vib config & in/out
				C	chamber reverb wet/dry
				D	flange feedback+4Tap mix (guitars)
				MW/SoftPd	rotary speed
				PSw2	*tap level*
845	World Rave Kit	132	GtRbSwpFlt FlDly	B	gated reverb wet/dry
				C	*sweep filt wet/dry*
				D	gated reverb time
				E	flange delay level
				PSw2	toggle: gated reverb/*sweep filt*
846	Punch Gate Kit	154	RoomRoomSRS CmRv	B	room reverb level & aux reverb level; room reverb absorption cut
				C	compress+reverb level (hi-hat & snare)
				PSw2	*compressor release time, config*
847	Shadow Kit	155	RoomRoom Room	B	reverb levels
				C	aux room level (elec. drum kit C#6-G 9)
				PSw2	reverb boost
848	Fat Traps	7	RoomFlgEcho Hall	B	room reverb wet/dry
				C	flange wet/dry & feedback level
				D	hall reverb level
				PSw2	*room reverb time cut, flange tempo*
849	Generator Kit	158	EnhcSp4T Hall	B	hall reverb level
				C	3-band enhancer (in/out)
				D	tap delay wet/dry
				PSw2	hall reverb time, EQ, echo length, high-frequency damp
850	Shudder Kit	75	HallPtchLsr Hall	B	aux hall reverb level, room size
				C	pitcher wet/dry
				D	hall reverb wet/dry
				E	Pitcher pitch
				PSw2	toggle: Pitcher/*LaserVerb*

Program ID	Program Name	Studio ID	Studio Name	Controller Assignments	
851	Crowd Stomper	154	RoomRoomSRS CmRv	B	FX1 reverb wet/dry, aux reverb wet/dry & time
				C	FX1 aux level & predelay, *FX2 reverb time*
				PSw2	toggle: room1/*room2 reverbs*
852	Econo Kit	38	RoomCmpCh4T Hall	B	hall reverb level & time
				C	room reverb wet/dry & time
				PSw2	toggle: compressor/*chorus+4Tap*
853	EDrum Kit 1	135	ChDlDstEQ Hall	B	hall reverb level
				C	distortion wet/dry
				D	chorus/delay wet/dry
				E	hall high-frequency damp, late reverb time
854	EDrum Kit 2	154	RoomRoomSRS CmRv	B	reverb levels
				C	aux reverb level
				PSw2	toggle room reverb/*SRS*
855	Dog Chases Tail	57	auxDistLasr Acid	B	reverb level (FX2)
				C	reverb level (FX3)
				D	reverb level (FX1)
				E	*LaserVerb wet/dry*
				PSw2	in A: distortion *in*/out; in B: toggle: distortion & *LaserVerb*
856	Saw Loop Factory	123	FlgEnv4Tap Plate	Data	Filter threshold, frequency & EQ
				B	reverb level
				C	env filt wet/dry
				D	filt resonance
				PSw2	toggle: env filt/*4Tap, EQ*
857	Two Live Bass	61	CompEQmphCh Room	B	room reverb level
				C	comp ratio
				D	*EQMorph panning*
				GAttVel	*EQMorph config*
				PSw2	toggle: compressor/*EQMorph*
858	Dual/Tri Bass	61	CompEQmphCh Room	B	room reverb level
				C	comp ratio
				D	EQ treble boost
				E	*room reverb level*
				F	*EQ gain*
				G	*EQ frequency scale*
				PSw2	toggle: comp/*EQmorph*
859	Clav-o-Bass	58	EnhcManPhs Room	B	room reverb level
				C	notch control
				D	*phaser LFO rate*
				PSw2	*phaser feedback boost*
860	Chirp Bass	130	auxEnvSp4T GtVrb	B	gated reverb level
				C	env filt wet/dry
				D	env filt attk rate
				E	*gated reverb level*
				F	*delay wet/dry*
				MWheel	env filt frequency sweep
				MPress	env filt resonance
				PSw2	toggle: env filt/*delay*
861	DigiBass	69	auxPtchDst+ Chmb	B	chamber reverb level
				C	pitcher wet/dry
				D	pitcher pitch
				E	odd wts
				F	pitch offset LFO
				F	*chamber reverb level*
				G	*distortion level*
				MPress	Pitcher pair wts.
				PSw2	toggle: pitcher/*distortion+*
862	Mono Synth Bass	57	auxDistLasr Acid	B	reverb level
				C	distortion wet/dry
				D	distortion drive
				E	*LaserDelay time*
				PSw2	toggle: distortion/*LaserDelay*

Program		Studio		Controller Assignments	
ID	Name	ID	Name		
863	Touch MiniBass	23	RmSweepEcho Hall	B	hall reverb level
				C	sweep filt wet/dry
				D	sweep filt LFO period
				E	sweep filter phase
				F	sweep filter LFO amplitude min frequency
				G	sweep filter LFO amplitude max frequency
				PSw2	toggle: sweep filt/*echo*
864	Ostinato Bass	62	BthQFlg4Tap Hall	B	hall reverb level
				C	booth reverb wet/dry
				D	*quantization+flange wet/dry & mix*
				E	*flange wet/dry*
				F	*flange feedback*
				PSw2	toggle: booth/aux hall & *quantization+flange*
865	House Bass	77	HallChorFDR Room	B	hall reverb wet/dry, *room reverb level*
				C	*chorus wet/dry*
				PSw2	toggle: hall reverb/*chorus*
866	Dubb Bass bad	90	auxPhsrFDR Hall	B	hall reverb level
				C	*phaser LFO depth*
				D	*phaser LFO rate*
				PSw2	vib phaser *in*/out
867	Straight Strat	6	RoomFlngCDR Hall	B	hall reverb level & high-frequency damp
				C	CDR wet/dry
				PSw2	toggle: CDR/*room reverb*
868	Chorus Gtr	63	ChmbTremCDR Room	B	room reverb level
				C	*CDR wet/dry*
				D	*CDR reverb mix*
				E	*CDR chorus mix*
				F	*CDR delay mix*
				PSw2	tremolo/*CDR*
869	Strataguitar	101	auxFlLsr SwHall	B	hall reverb level
				C	LaserVerb wet/dry
				PSw2	flange in/*out*, EQ, *LaserVerb config*
870	Elect 12 String	39	RmDsRotFl4t RvCm	B	reverb+comp level
				C	flange mix
				D	flange tempo
				E	flange Xcursion
				F	tap delay mix
				G	flange+4T wet/dry, out gain
				MW	rotor rate
				PSw2	toggle: rotary+distortion/*flng+4Tap*
871	Dyn Jazz Guitar	101	auxFlngLasr Hall	B	hall reverb level
				C	hall reverb time
				D	*flange wet/dry*
				E	*flange LFO tempo*
				F	*flange feedback level*
				PSw2	*flange in/out*
872	Pedal Steel	101	auxFlngLasr Hall	B	reverb level, time, high-frequency damp
				D	*flange feedback level*
				E	*flange LFO tempo*
				PSw2	*adds flange*
873	Strummer DistGtr	94	auxChorMDly Hall	B	hall reverb level
				C	delay wet/dry
				PSw2	chorus in/*out*
874	Rock Axe	93	auxChrDst+ Hall	B	delay wet/dry, hall reverb level
				C	chorus feedback level
				D	chorus rate
				E	chorus depth (left channel)
				PSw2	*distortion EQ*, chorus *in*/out
875	Hammeron	16	RoomPhsrCDR Hall	B	hall reverb level
				C	delay level
				PSw2	toggle: CDR/*room*

Program ID	Program Name	Studio ID	Studio Name	Controller Assignments	
876	Rock Axe mono	93	auxChrDst+ Hall	B	delay level, reverb level
				C	*distortion+chorus wet/dry*
				D	*distortion+chorus feedback level*
				E	*distortion+chorus rate*
				F	*distortion+chorus depth*
				PSw2	toggle: chorus/*distortion+chorus+delay*
877	Attack Stack	84	HallFlgChDl Hall	B	reverb levels, times
				C	high-frequency damp, EQ boost
				PSw2	toggle: hall/*flange*
878	Skinny Lead	137	AuxChorFlng CDR	B	CDR level, reverb time
				C	flange wet/dry & feedback level, treble cut
				D	CDR chorus feedback
				E	flange LFO tempo
				G	CDR delay tempo & feedback
				PSw2	flange LFO1 phase, CDR chorus rate cut, EQ
879	Q Sweep SynClav	137	AuxChorFlng CDR	B	CDR level, reverb time
				C	chorus wet/dry, bass cut
				D	chorus feedback & Xcouple
				E	CDR delay mix
				F	CDR delay tempo
				G	CDR delay feedback
				H	CDR delay wet/dry
				PSw2	toggle: chorus+CDR/*flange*
880	Anna Mini	13	RmFlgFXFlng Flng	B	flange levels
				C	FX2 flange tempo & level
				D	FX2 flange feedback level
				E	EQ bass boost
				F	aux flange wet/dry & feedback level
				G	aux flange LFO tempo
				PSw2	toggle: “Delirium” & “*Throaty*” flanges
881	Ballad Stack	29	RoomSrsCDR CDR	B	aux CDR level
				C	aux CDR chorus feedback level
				D	aux CDR delay feedback & mix level
				E	aux CDR chorus rate
				F	aux CDR delay tempo
				G	SRS center frequency cut, space boost
				PSw2	toggle: SRS/*CDR*
882	Big Stack	85	Hall Room SRS	B	hall reverb wet/dry
				C	SRS level
				D	SRS center/space, EQ lo & hi boost
				PSw2	hall in/out, *EQ*
883	BrazKnuckles	85	Hall Room SRS	B	hall reverb wet/dry & decay time
				C	SRS level
				D	SRS center/space
				E	SRS EQ boost
				PSw2	hall reverb in/*out*, EQ, *SRS panning*
884	Hybrid Breath	140	EnhcChorChDl PCD	B	PCD chorus feedback level
				C	PCD delay feedback & mix level
				D	PCD level
				PSw2	chorus *in*/out
885	Hybrid Stack	13	RmFlgFXFlng Flng	B	reverb wet/dry & quality
				C	aux flange level
				D	aux LFO tempo
				E	aux flange wet/dry & feedback level
				PSw2	toggle: room+aux flange/*flange*
886	Eye Saw	13	RmFlgFXFlng Flng	B	aux flange level, EQ
				C	flange wet/dry
				D	flange feedback level
				E	aux flange wet/dry & feedback level
				F	aux LFO tempo
				G	flange Xcursion, LFO tempo & Xcouple

Program ID	Program Name	Studio ID	Studio Name	Controller	Controller Assignments
887	Mello Hyb Brass	3	RoomChorCDR Hall	B	room & hall reverb level, room wet/dry
				C	chorus feedback level
				D	chorus Xcouple
				GAttVel	EQ bass boost
				PSw2	toggle: room & *chorus*
888	Sizzl E Pno	97	auxPhasStIm Hall	B	hall reverb level, time, & high-frequency damp
				C	phaser wet/dry
				D	phaser LFO rate
				E	*hall reverb level*
				F	*EQ, stereo image spread & ctr gain*
				PSw2	toggle: phaser/*stereo image*
889	My JayDee	8	RmFlngStImg Garg	B	reverb level
				C	reverb high-frequency damp (all)
				PSw2	toggle: room reverb/*flange*
890	Slo SynthOrch	97	auxPhasStIm Hall	B	hall reverb wet/dry & time
				C	EQ boost, stereo image in gain
				D	hall reverb early reflection boost, late real cut
				PSw2	*stereo image mix*
891	SpaceStation	8	RmFlngStImg Garg	B	EQ mod
				F	flange feedback level
				G	flange LFO tempo, garage reverb level
				H	garage reverb wet/dry
				PSw2	*stereo image mix*
892	Glass Web	152	auxFlgDst+ ChLsD	B	aux chorus/delay level, flange LFO tempo, aux chorus mix & feedback
				C	flange feedback
				D	aux chorus/LaserDelay wet/dry
				E	aux delay feedback
				F	aux delay tempo
				G	flange wet/dry & Xcurs, aux chorus rate
893	Circus Music	151	ChDlSp4TFlDl Phs	B	4Tap wet/dry
				C	4Tap feedback level
				D	phaser level, 4Tap mix level
				E	4Tap feedback image
				F	phaser feedback
				G	phaser notch/bandpass
				H	4Tap delay tempo
				MWheel	phaser rate
				GKeyNum	4Tap pitch adjust
894	Mandala	151	ChDlSp4TFlDl Phs	B	phaser level (koto)
				C	4Tap wet/dry & feedback (koto)
				D	4Tap feedback image
				E	phaser feedback
				F	4tap delay tempo
				GKeyNum	4Tap pitch adjust
				MWheel	phaser rate
895	Slow Strat	136	auxDPanCDR ChPlt	B	aux chorus/plate reverb level
				C	panner LFO rate & pulse width
				D	aux chorus feedback
				E	aux chorus depth
				F	aux chorus Xcouple
896	Fluid Koto	151	ChDlSp4TFlDl Phs	B	phaser level, EQ
				C	tap delay wet/dry & feedback
				D	tap delay feedback image
				E	phaser feedback
				H	tap delay tempo
				GKeyNum	tap delay pitch adjust
				MW	aux phaser center frequency
897	Koreana Pad	134	ChorChorCDR Spac	B	space reverb level, tap chorus wet/dry
				C	tap chorus feedback
				D	tap chorus LFO rate
				E	chorus feedback level
				PSw2	toggle: tap chorus/*chorus*

Program		Studio		Controller Assignments	
ID	Name	ID	Name		
898	Tangerine	140	EnhcChorChDl PCD	B	PCD chorus feedback, enhancer mid & lo drive
				C	PCD delay mix & feedback
				D	PCD level
899	Planet 9	137	AuxChorFlng CDR	B	CDR level & reverb mix & time
				C	flange wet/dry & feedback, EQ
				D	CDR chorus feedback
				E	flange LFO tempo
				F	flange LFO phase
				G	CDR delay tempo & feedback

Setup Control Assignments

Setup ID	Setup Name	Studio ID	Studio Name	Controller	Assignment
800	HyperGroov<-C4->	112	PlatEnvFl4T Filt	E	filter type
				F	filter level
				G	reverb wet/dry & quality; flange feedback level
801	PianoPad w/Percs	74	HallFlgChDl Room	E	filter flange feedback
				F	flute & percussion reverb level
				G	piano reverb wet/dry
802	Slo Held Arper	6	RoomFlngCDR Hall	G	piano/vox reverb wet/dry & delay level
803	Don'tGetFooled	25	RmRotoFl4T CmpRv	F	Flange level
				G	aux reverb wet/dry
				H	4-Tap level
				PSw1	Arpeggiator in/out
				PSw2	*vib/chorus* in/out
				MW/SoftPd	rotor rate
804	Touch Game	114	PltTEnvFlg Plate	F	perc reverb wet/dry & env filter expression
				G	comp reverb wet/dry & env filter expression
805	BeatBoy E1	67	ChmbEnv4Tap GtRv	E	kick/snare gate time
				F	pad-under-lead flamdelay wet/dry
				G	aux reverb wet/dry
806	ZawiClav Split	92	auxFlgDist+ Hall	G	lead MDdelay/ feedback
				MPress	lead tube drive
807	Dyn Piano Pad	159	Room RoomChr SRS	F	SRS center/space EQ level
				G	SRS reverb wet/dry
808	Pulsar Stack	153	auxFlgDst+ ChLs2	D	lead-pad flange level/feedback
				E	lead-pad hi-frequency damp
				F	lead-pad delay color
				G	lead-pad flange gain/LFO Tempo
809	Mt Chicorora C2	71	auxChorFlRv Cmb2	G	perc reverb time
				MWheel	pad bass boost
810	Hold Low 3sec Rb	78	HallPtchPtFl Lsr	Data	bass & lead LaserVerb feedback level
				G	bass & lead LaserVerb wet/dry
				SmRbn	slithery alien effect
				Tempo	bass & lead delay & pitch
811	Mettlorfus Pad	69	auxPtchDst+ Chmb	E	perc pitch level
				F	perc reverb
				G	lead drive outgain level
				LgRbn	perc pitch quality
				MPress	lead drive crunch
812	Black Keys xtra	6	RoomFlngCDR Hall	E	kit Flange level
				F	Perc chorus+delay+reverb level
				G	kit reverb level & perc (Zone 1) reverb wet/dry
813	Jungle Jammer	23	RmSweepEcho Hall	F	right-hand perc sweep filter level
				G	right-hand perc reverb wet/dry
814	Huge Rock Band	25	RmRotoFl4T CmpRv	E	lead reverb wet/dry, band delay level
				F	aux reverb wet/dry
				G	rotor trigger
				PSw1	zone mutes
				PSw2	vib in/out
815	Rock Ballad	39	RmDsRotFl4t RvCm	F	distorted gtr flange level
				G	kit reverb time
				H	aux reverb wet/dry
				SoftPd	rotor trigger
816	Jazz Setup	94	auxChorMDly Hall	E	lead delay level & feedback
				F	bass chorus wet/dry
				G	reverb level
817	Two Touchers	94	auxChorMDly Hall	E	right-hand lead delay wet/dry
				F	right-hand lead reverb level
				G	left-hand comp reverb level
818	Frontier prs	23	RmSweepEcho Hall	G	pad reverb level

Setup ID	Setup Name	Studio ID	Studio Name	Controller Assignments	
819	Electric Grand	43	Room Room Hall	E	piano1 reverb wet/dry
				F	piano2 reverb wet/dry
				G	hall reverb level
820	Bad Trip FtSw/MW	55	auxDistLasr Room	F	LaserDelay time
				G	room reverb level
821	WhirliToys	90	auxPhsrFDR Hall	E	(Zones 1, 3, 7) flange level & feedback
				F	(Zones 1, 3, 7) delay level; flange + delay wet/dry
				G	hall reverb level
822	PluckSynths Perc	72	auxChorFlRv Cmb3	F	fluty synth orch flange level
				G	chamber reverb level
823	SusPed RhythmJam	68	CmbrShapLsr Hall	F	lead LaserVerb wet/dry
				G	aux reverb wet/dry & chamber wet/dry
				GAttVel	lead LaserVerb delay time/contour
				Sustain	comp shaper intensity
824	Ballad Piano Pad	82	HallRsFltChDl Rm	F	pad resonant filter wet/dry
				G	pad reverb send
				GKeyNum	bass EQ frequency
				Sustain	filter sweep ASR
825	Big AnaLoveVibe	63	ChmbTremCDR Room	G	room reverb level; CDR wet/dry
				GAttVel	stack panning tremolo rate/depth
826	ShockBreaks Psw1	17	RmPhsrQuFlg Hall	F	flange wet/dry, feedback level
				G	hall reverb level
				PSw1	quantization distortion effect
827	Four Pluckers	75	HallPtchLsr Hall	E	LaserDelay coarse
				F	LaserDelay fine
				G	aux reverb level; LaserDelay spacing
				H	LaserDelay contour
				GKeyNum	pitch tracking
828	WaterPiano Pad	56	auxEnhSp4T Class	F	pad delay wet/dry
				G	lead reverb level
829	Padded Room	94	auxChorMDly Hall	F	lead delay wet/dry
				G	hall reverb level
830	AtmosPolySphere	90	auxPhsrFDR Hall	G	pad flange/delay/reverb wet/dry
831	Breath Pad	63	ChmbTremCDR Room	G	lead delay wet/dry, feedback, high-frequency damp
				MPress	pad tremolo Tempo, room reverb level
832	Trippy Jam	74	HallFlgChDl Room	F	organ flange feedback
				G	bell-lead room reverb level; organ flange feedback
				GAttVel	bell-lead delay mix level
833	MeditationGuits	63	ChmbTremCDR Room	F	lead chorus mix level
				G	lead reverb wet/dry, room reverb level, delay feedback
834	Cool Down Funk	137	auxChorFlng CDR	F	clav flange wet/dry & excursion; CDR delay wet/dry
				G	CDR reverb level & E Piano treble boost
835	Tek`Groov C5->	128	Gtd2ChrEcho 2Vrb	F	bass reverb level
				G	kits reverb level
836	Big Fat Split	6	RoomFlngCDR Hall	F	bass hall reverb level
				G	lead delay mix, hall reverb level
837	The Pump C2	21	RmEQmph4Tp Space	D	kit EQ frequency and morph
				E	kit delay wet/dry
				F	kit aux reverb level
				G	kick, snare, bass aux reverb level
838	Ana Basses	62	BthQFlg4Tap Hall	F	lead quantize-flange wet/dry
				G	lead hall reverb level
839	Multi Followers	33	ChmbCompCDR Hall	F	pad delay
				G	room & hall reverb level
840	Plucksynths	6	RoomFlngCDR Hall	F	pad chorus rate, quality
				G	lead reverb wet/dry, time; mix hall reverb level
841	10 Leagues Under	90	auxPhsrFDR Hall	G	pad hall reverb level, FDR wet/dry
				Chan S	pad treble boost, phaser wet/dry
842	Gremlin Arps	75	HallPtchLsr Hall	G	arp pitcher & LaserVerb wet/dry
				MPress	pitcher LFO rate
843	Broken Toys	76	HallGateFl4T Bth	F	booth reverb level
				G	delay depth
844	Two Synth	33	ChmbCompCDR Hall	G	hall reverb level, pad hi boost, piano lo boost

Setup		Studio		Controller Assignments	
ID	Name	ID	Name		
845	Machine Shop	17	RmPhsrQuFlg Hall	D	kit1 phaser wet/dry
				E	kit2 quantize + flange wet/dry
				F	lead reverb wet/dry
				G	hall reverb level
				Tempo	hall reverb space, phaser rate
846	Farawaway Place	90	auxPhsrFDR Hall	F	pad hall reverb level
				G	organ hall reverb level
847	BehindEnemyLines	91	auxChrDist+ Hall	G	hall reverb level, MDdelay wet/dry
848	Tunnel Visionprs	6	RoomFlngCDR Hall	E	flange wet/dry
				F	CDR wet/dry
				G	hall reverb level
				Chan S	treble boost & fade
849	Seismic Trance	132	GVrbSwpFlt DlyFl	E	kit gateverb wet/dry
				F	kit gate threshold level
				G	delay + flange wet/dry, sweep filter wet/dry
850	Medal	74	HallFlgChDl Room	E	pad chorus/delay wet/dry
				F	brazz level
				G	brazz reverb level

Controller Assignments: Contemporary ROM Block

Appendix E
Orchestral ROM Block Objects

This Appendix describes the Orchestral ROM objects provided with your K2661.

Programs

Orchestras	
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 K2661 Orchestral ROM 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
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	
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	

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
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	
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	

Prg ID	Program Name	Mod Wheel	Data	MPress	Comments
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	

Controller Assignments: Orchestral ROM Block

This section lists the controller assignments for all programs and setups in the Orchestral ROM sound block.

Secondary Effects

Some of the programs in the Orchestral block use a programming technique called *secondary effects*, in which the processing on one or more layers of the program can be changed with the press of a button. Secondary effects in these programs are enabled by PSw2 (or by any physical controller assigned to send MIDI 29). PSw2 acts as a toggle between the primary effect and the secondary effect. It switches off one of the two FXBus sends on an Input page (sets its Lvl parameter to **Off**), and simultaneously turns on the other FXBus send (sets its Lvl parameter to **0.0 dB**).

The following diagram shows the effect of pressing PSw2 on the settings for FXBus1 and FXBus2.

PSw2 Status	Value of Lvl Parameter on Input Page	
	FXBus1	FXBus2
Off	0.0 dB	Off
On	Off	O.0 dB

In most cases, toggling effects with PSw2 affects only a single layer on a single input pair. In some cases, however, the switching is more complicated, and toggling effects moves one or more layers to different FX buses. Toggling effects may also change EQ settings, or the Aux reverb's decay time, depending on the program.

The following segment from the controller listings shows an example of secondary effects. Secondary effects appear in italics. In this example, when PSw2 is off, the program's input routings result in a room reverb effect, Slider B controls the wet/dry mix of this reverb. When PSw2 is on, the routing changes, resulting in a flange effect. In this case, Slider B is inactive, Slider C controls the aux room reverb level, and Slider D controls both the flange level and the crosscouple amount.

Program		Studio		Controller Assignments	
ID	Name	ID	Name		
999	SuperSynth	9	RmFlgChDly Room	B	room1 reverb wet/dry
				C	*aux room reverb level*
				D	*flange level, flange Xcouple*
				PSw2	toggle: room1 reverb/*flange*

Program Control Assignments

Program		Studio		Controller Assignments	
ID	Name	ID	Name		
900	TotalCntrl Orch1	110	Chapel Room Hall	B	room, hall, & chapel reverb time
				C	chapel level
				PSw2	toggle room reverb
901	TotalCntrl Orch2	110	Chapel Room Hall	B	room, hall, & chapel reverb level & time
				PSw2	toggle chapel
902	Baroque Orchestra	110	Chapel Room Hall	B	room, hall, and chapel reverb level & time
				PSw2	toggle chapel
903	Oboe&Flute w/Str	108	ChapelSRS Hall	B	chapel reverb wet/dry & time
904	Horn&Flute w/Str	110	Chapel Room Hall	B	room & hall reverb level, room reverb time
905	Trp&Horns w/Str	110	Chapel Room Hall	B	room & hall reverb level
				PSw2	decreases reverb time
906	Piccolo	42	RoomRmHall Hall	B	aux hall reverb level & time, room reverb wet/dry
				PSw2	decreases aux hall brightness
907	Orchestral Flute	42	RoomRmHall Hall	B	aux hall reverb level & time
				PSw2	increases room (FX1) time
908	Solo Flute	42	RoomRmHall Hall	B	aux hall reverb level & time, room reverb time
				PSw2	decreases aux hall brightness
909	Orchestral Oboe	42	RoomRmHall Hall	B	aux hall reverb level & time
				PSw2	decreases aux hall brightness and room (FX1) time
910	Solo Oboe	42	RoomRmHall Hall	B	aux hall reverb level & time
				PSw2	decreases aux hall brightness and room (FX1) time
911	2nd Oboe	42	RoomRmHall Hall	B	aux hall reverb level
912	Orch EnglishHorn	42	RoomRmHall Hall	B	aux hall reverb level & time
913	Solo EnglishHorn	42	RoomRmHall Hall	B	aux hall reverb level & time
914	Orch Clarinet	42	RoomRmHall Hall	B	aux hall reverb level & time
915	Solo Clarinet	42	RoomRmHall Hall	B	aux hall reverb level & time
916	Orch Bassoon	42	RoomRmHall Hall	B	aux hall reverb level & time
917	Solo Bassoon	42	RoomRmHall Hall	B	aux hall reverb level & time
918	Woodwinds 1	42	RoomRmHall Hall	B	aux hall reverb level & time
919	Woodwinds 2	42	RoomRmHall Hall	B	aux hall reverb level & time
920	Dynamic Trumpet	34	RoomCmpChor Hall	B	room & hall reverb level & time
921	Copland Sft Trp	42	RoomRmHall Hall	B	aux hall reverb level
922	Orch Trumpet	42	RoomRmHall Hall	B	aux hall reverb level, room reverb time
923	Soft Trumpet	42	RoomRmHall Hall	B	aux hall reverb level
924	Strght Mute Trp	35	RoomComp Hall	B	aux hall reverb level
925	French Horn MW	44	Room Hall Hall	B	aux hall reverb level, room reverb time
926	Slow Horn	44	Room Hall Hall	B	aux hall reverb level, room reverb time
927	F Horn Con Sord	44	Room Hall Hall	B	aux hall reverb level & time, room reverb time
928	F Horn a2 MW	44	Room Hall Hall	B	aux hall reverb level, room reverb time
				MWheel	aux hall time
929	French Horn Sec	44	Room Hall Hall	B	aux hall reverb level, room reverb time
930	French Horn Sec2	44	Room Hall Hall	B	aux hall reverb level, room reverb time
931	Solo Trombone	44	Room Hall Hall	B	aux hall reverb level, room reverb time
932	Tuba	44	Room Hall Hall	B	room & aux hall reverb level
933	Dyn Hi Brass	42	RoomRmHall Hall	B	room (FX1) time & aux hall reverb level
				C	*room (FX2) wet/dry*
				D	*room (FX2) high-frequency damp*
				E	*room (FX2) time*
				PSw2	toggle room (FX1) and *room (FX2)*
934	Dyn Lo Brass	44	Room Hall Hall	B	aux hall reverb level, room reverb time
				C	aux hall high-frequency damp
				PSw2	toggle room
935	Dyn Brass & Horn	44	Room Hall Hall	B	aux hall reverb level & *room reverb time*
				MWheel	room reverb roll-off
				PSw2	toggle room
936	Soaring Brass	44	Room Hall Hall	B	aux hall reverb level & time
937	MarcatoViolin MW	35	RoomComp Hall	B	room & hall reverb level
938	Solo Violin	35	RoomComp Hall	B	room & hall reverb level

Program		Studio		Controller Assignments	
ID	Name	ID	Name		
939	2nd Violin	35	RoomComp Hall	B	hall reverb level
				C	room level
940	Orch Viola	35	RoomComp Hall	B	room & hall reverb level
941	Solo Viola	35	RoomComp Hall	B	room & hall reverb level
942	Slow Viola	35	RoomComp Hall	B	hall reverb level
943	MarcatoCello MW	35	RoomComp Hall	B	room & hall reverb level
944	Solo Cello	35	RoomComp Hall	B	room & hall reverb level
945	Slow Cello	35	RoomComp Hall	B	room & hall reverb level
946	Arco Dbl Bass	35	RoomComp Hall	B	hall reverb level
				C	room level
947	Slow Arco Bass	35	RoomComp Hall	B	room & hall reverb level
948	Brt Dbl Bass	35	RoomComp Hall	B	room & hall reverb level
949	Touch Strings	86	Hall Room Room	B	hall reverb wet/dry & time
950	Fast Strings MW	86	Hall Room Room	B	hall reverb wet/dry & time
951	Chamber Section	86	Hall Room Room	B	hall reverb time
952	Sfz Strings MW	86	Hall Room Room	B	hall reverb wet/dry & time
953	Sweet Strings	86	Hall Room Room	B	hall reverb wet/dry & time
954	Baroque Strg Ens	86	Hall Room Room	B	hall reverb wet/dry & time
955	Big String Ens	86	Hall Room Room	B	hall reverb wet/dry & time
956	Bass String Sec	86	Hall Room Room	B	hall reverb wet/dry & time
957	Pizzicato String	86	Hall Room Room	B	hall reverb wet/dry & time,high-frequency damp
958	Wet Pizz	86	Hall Room Room	B	hall reverb wet/dry & time, high-frequency damp
959	Arco & Pizz	86	Hall Room Room	B	hall reverb wet/dry & time, high-frequency damp
960	Classical Guitar	108	ChapelSRS Hall	B	chapel reverb wet/dry & time
961	Virtuoso Guitar	108	ChapelSRS Hall	B	chapel reverb wet/dry & time
962	Acoustic Bass	108	ChapelSRS Hall	B	room reverb wet/dry
963	Snappy Jazz Bass	108	ChapelSRS Hall	B	room reverb wet/dry
964	Dynamic Harp	108	ChapelSRS Hall	B	chapel reverb wet/dry & time
965	Harp w/8ve CTL	108	ChapelSRS Hall	B	chapel reverb wet/dry & time
966	Harp Arps	108	ChapelSRS Hall	B	chapel reverb wet/dry & time
967	Celesta	108	ChapelSRS Hall	B	chapel reverb wet/dry & time
968	Pipes	108	ChapelSRS Hall	B	chapel reverb wet/dry
				C	hall reverb level
969	Pedal Pipes 2	108	ChapelSRS Hall	B	chapel reverb wet/dry
				C	hall reverb level
970	Church Bells	109	ChapelSRS Hall2	B	room & hall reverb level
971	Glockenspiel	108	ChapelSRS Hall	B	chapel reverb wet/dry & time
				C	hall reverb level
972	Xylophone	108	ChapelSRS Hall	B	chapel reverb wet/dry
				C	hall reverb level
973	Chimes	109	ChapelSRS Hall2	B	chapel reverb wet/dry
				C	hall reverb level
974	Timpani/Chimes	108	ChapelSRS Hall	B	chapel & hall reverb level & time
975	Timpani	108	ChapelSRS Hall	B	chapel reverb wet/dry
				C	hall reverb level
976	Timpani & Perc	110	Chapel Room Hall	B	chapel reverb wet/dry & time
				C	hall reverb level
977	Big Drum Corp	89	HallRoomChr Hall	B	reverb wet/dry
978	Orch Percussion1	100	auxSRSRoom Hall	B	hall reverb level
				C	dry level cut
979	Orch Percussion2	100	auxSRSRoom Hall	B	hall reverb level
980	Jam Corp	89	HallRoomChr Hall	B	reverb wet/dry
				C	reverb absorption amount
981	Conga & Perc	45	Room Room Hall2	B	room reverb wet/dry
				C	hall reverb level
982	Woody Jam Rack	37	BthComp SRS Hall	B	reverb wet/dry
				C	reverb absorption amount
983	Metal Garden	62	BthQFlg4Tap Hall	B	booth reverb wet/dry & absorption amount
				C	hall reverb level
984	Hot Tamali Kit	38	RoomCmpCh4T Hall	B	room reverb wet/dry & time
				C	hall reverb level & time
				D	high-frequency damp level
985	Funk Kit	158	EnhcSp4T Hall	B	aux reverb level

Program ID	Program Name	Studio ID	Studio Name	Controller	Assignment
986	Magic Guitar	3	RoomChorCDR Hall	B	hall reverb level
				C	*chorus+delay+reverb wet/dry*
				D	*reverb wet/dry*
987	Glass Bow 2	26	RoomSrsCDR Hall	B	hall reverb level
988	Synth Orch	52	auxChrMDly Room	B	room reverb level
				C	room reverb time
				D	LFO depth
				SostPd	infinite decay i/o
989	Nooage InstaHarp	102	auxEnh4Tap Hall	B	hall reverb level
990	AC Dream	121	auxMPFlgLasr Plt	B	reverb level
991	Synth Dulcimer	40	RoomRmHall Hall	B	aux hall reverb level
992	Glistener	113	PltEnvFl4T Plate	B	aux plate reverb level
				C	flange + delay wet/dry
993	Afro Multi CTL	129	GtdEnhcStIm Hall	B	hall reverb level
				C	gate reverb wet/dry
994	Tranquil Sleigh	74	HallFlgChDl Room	B	room reverb level
				C	flange wet/dry
995	Batman Strings	11	RoomFlngCDR Hall	B	Batcave reverb level
				C	flange wet/dry
996	Ethnoo Lead	119	auxChorDist+ Plt	B	plate reverb level
				C	chorus wet/dry
				D	tube drive level
				E	MD delay wet/dry
				F	MD delay time
				G	MD delay feedback
997	Orch Pad CTL	66	ChamFlg4Tap Hall	B	room & hall reverb level
				C	hall reverb decay time
				D	EQ bass boost
				E	EQ treble boost
998	Choral Sleigh	2	RmChorChRv Hall	B	aux hall reverb level, *voice aux level*
				C	voice room reverb wet/dry
				MWheel	pad chorus wet/dry, *voice chorus wet/dry*
				PSw2	toggles room & *chorus*
999	Pad Nine	98	auxFlngCDR Hall	B	hall reverb level
				C	hall reverb time
				D	hall reverb level
				F	*flange wet/dry*
				G	*flange feedback level*
				PSw2	*toggle flanger*

Setup Control Assignments

Setup		Studio		Controller Assignments	
ID	Name	ID	Name		
900	Deep Piano Rbn	16	RoomPhsrCDR Hall	G	CDR wet/dry, pad & piano hall reverb level
901	Choir & Harp	42	RoomRmHall Hall	E	room wet/dry & time
				F	choir hall reverb time
				G	all zones (aux) hall2 level
902	Orchestrator	133	ChRvStIEcho Hall	G	chorus/reverb wet/dry
903	Piano Concerto	42	RoomRmHall Hall	E	woodwinds and brass reverb wet/dry
				F	strings and perc reverb wet/dry
				G	aux reverb level
				H	piano reverb wet/dry
904	Xmas Carols	44	Room Hall Hall	E	brass room reverb wet/dry
				F	chimes and timpani hall reverb wet/dry
				G	all zones hall2 reverb level
905	Sideline Perc	89	HallRoomChr Hall	F	drums and perc chorus wet/dry
				G	reverb level
906	TonalGroov C5->	34	RoomCmpChor Hall	G	hall reverb level
907	Exotic Grooves	149	auxPtchRoom RvCm	G	perc aux reverb level
908	Lunar Harp	133	ChRvStIEcho Hall	G	pad & harp chorus/reverb wet/dry, harp hall reverb level
909	Themes	77	HallChorFDR Room	F	choir chorus wet/dry
				G	room reverb level
910	Wet Piano	42	RoomRmHall Hall	F	piano distance
				G	hall reverb level; flute room reverb level
				H	piano lead reverb wet/dry room
911	enter the Jester	42	RoomRmHall Hall	G	reverb level & time
912	Tap the Jester	42	RoomRmHall Hall	G	reverb level & time
913	Hybrid Strings	42	RoomRmHall Hall	F	pad reverb wet/dry
				G	aux reverb level
914	Wonderous Spaces	74	HallFlgChDl Room	F	harp delay mix wet/dry
				G	room reverb level
915	Metal Orch Pad	11	RoomFlngCDR Hall	G	hall reverb level & time
916	Toon prs	42	RoomRmHall Hall	G	aux reverb level
917	Tranquil Sea	11	RoomFlngCDR Hall	G	hall reverb level
918	Sick Clock Jam	149	auxPtchRoom RvCm	G	bell aux reverb level
				GAttVel	bass reverb/compressor level
919	Orc Split	26	RoomSrsCDR Hall	G	reverb level
920	Baroque Brass	45	Room Room Hall2	G	hall2 reverb level
921	Unison Orchestra	45	Room Room Hall2	G	hall2 reverb level
922	Unison w/Pizz	45	Room Room Hall2	G	hall2 reverb level
923	Switch Orchestra	100	auxSRSRoom Hall	G	hall reverb level
924	Pizz/Str/Winds	2	RmChorChRv Hall	G	aux reverb level
925	Harp Arps Cmaj	121	auxMPFlgLasr Plt	G	plate reverb level
926	Desert Bloom E1	6	RoomFlngCDR Hall	G	string pad flange wet/dry
927	Exotic Charge	33	ChmbCompCDR Hall	F	pad delay mix wet/dry
				G	reverb level
928	ET Comes Home	129	GtdEnhcStIm Hall	G	hall reverb level
929	Fanfare Orch	1	RoomChorDly Hall	E	delay mix wet/dry, chorus feedback level
				F	chorus mix wet/dry
				G	hall reverb wet/dry & delay wet/dry
930	Switch Orch 2	1	RoomChorDly Hall	E	delay mix wet/dry
				F	chorus mix wet/dry
				G	reverb level & delay wet/dry
931	Orbiting Venus	80	HallChrEcho Room	E	echo feedback image
				F	chorus wet/dry & feedback
				G	echo wet/dry & high-frequency damp reverb wet/dry
				H	echo feedback level
932	Glass Dulcimer	81	HallChorCDR Hall	E	CDR delay mix level; chorus feedback level
				F	chorus wet/dry
				G	pad reverb wet/dry
				H	delay mix level, chorus feedback level

Setup ID	Setup Name	Studio ID	Studio Name	Controller	Assignment
933	Hybrid Reeds	1	RoomChorDly Hall	E	lead delay mix
				F	lead chorus mix
				G	reverb & effects wet/dry
934	Two Hand Pizz	1	RoomChorDly Hall	G	reverb wet/dry
				GAttVel	bass cut
935	Slo Str & Horn	47	Room Room Hall2	G	reverb wet/dry
936	Pianist Band	159	Room RoomChr SRS	F	drums reverb wet/dry
				G	piano reverb wet/dry & time
				H	*SRS center/space*
				PSw2	SRS in/out
937	Prepared Pianos	16	RoomPhsrCDR Hall	E	toggles reverb delay effect
				F	toggles reverb density effect
				G	room1 reverb wet/dry, time, high-frequency damp, diffusion
938	FSW1 solo winds	47	Room Room Hall2	E	pad reverb wet/dry
				F	pad hall2 reverb level
				G	lead hall2 reverb level
939	Strings&Winds	47	Room Room Hall2	E	winds reverb wet/dry
				F	winds hall2 reverb level
				G	strings hall2 reverb level
940	Str Ens Solo MW	48	Room Hall Hall2	F	room reverb level
				G	hall2 reverb level
				MWheel	treble EQ gain
941	Pno&Vox&Pizz	31	RoomSRSRoom Chmb	F	room1 & room2 reverb wet/dry
				G	chamber reverb level
942	Down Wind SmRbn	5	RoomChrCh4T Hall	G	reverb & chorus & delay wet/dry
				MWheel	wind chorus LFO rate
943	Gtr & Piano	134	ChDlyChrCDR Spac	D	acoustic guitar delay mix, piano chorus wet/dry
				E	electric guitar chorus wet/dry
				F	electric guitar chorus feedback
				G	acoustic guitar reverb wet/dry, electric guitar chorus depth
				H	acoustic guitar chorus mix, electric guitar & piano rates
944	Cirrus 9	103	EnhcChorCDR Hall	E	hall reverb level & enhancer high drive
				F	pad chorus wet/dry & chorus rate
				G	hall reverb space, pad chorus feedback
945	Dry Plucks	5	RoomChrCh4T Hall	F	bass chorus wet/dry & feedback level
				G	piano reverb level
946	String Collage	32	RoomSRSRoom Hall	F	hall reverb time
				G	hall reverb level
947	Esoterica	107	ChorChorFlg Hall	F	“Cymbal Thing” level
				G	hall reverb level
948	Poseidon	59	EnhrFlg8Tap Room	D	pan balance
				E	pad EQ frequency & bass gain
				F	pad treble boost
				G	pad flange feedback
				H	pad flange LFO Tempo
949	Stalkers	138	auxEnhcSp4T CDR	F	CDR delay mix
				G	CDR reverb level
950	Diabolic Trickle	15	ChmbFlngCDR Verb	F	aux reverb level, pad chorus level, feedback, & rate
				G	bell reverb level, doom feedback

Appendix F
SD Piano ROM Option

SmartMedia Contents

The objects for ROM3 (SD Piano) are included on the SmartMedia card that comes with your K2661. The procedure for installing the objects is described in the installation instructions that came with your SD Piano option kit.

Sympathetic Vibrations

When you play a chord on an acoustic piano and hold down the keys while the notes decay, the dampers on the corresponding strings remain up, and you hear a particular set of harmonics that evolve from the undamped strings. You don't hear any significant harmonics from the other strings.

If you play the same chord and hold it with the sustain pedal, you'll hear a much different, richer set of harmonics as the notes decay. That's because *all* the strings are undamped, and each string gradually begins to vibrate at its resonant frequency, in response to the vibrations of the strings struck by the hammers.

This phenomenon is called sympathetic vibration, and is an important component of the sound of an acoustic piano. The most noticeable of these sympathetic vibrations come from the strings whose fundamental pitches match the harmonics of the strings that were struck by the hammers.

To create a more realistic acoustic piano sound for the SD Piano option, Kurzweil sound engineers have developed special effects settings that imitate sympathetic vibrations. When you're playing one of the SD Piano programs with ID 700–713, and you're not using the sustain pedal, the K2661's audio signal passes through FXBus 1, then to the AuxBus, which applies a typical room or hall reverb.

When you use the sustain pedal, the signal passes FXBus 1 *and* FXBus 2 before going to the AuxBus. FXBus 2 applies a chain of reverbs programmed to resemble an acoustic piano's sympathetic vibrations.

You can use the Data slider (or any physical controller sending MIDI 6) to adjust the level the sympathetic vibration effect.

Modifying SD Piano Programs

If you want to change the room ambiance on an SD Piano program without losing the sympathetic vibration effect, use the Studio Editor to assign a different effect on the AuxBus of the studio used by that program. If you want to remove the sympathetic vibrations, remove the effect from FXBus 2.

Controller Assignments for SD Piano Programs

The tables in this section list the controller assignments for the SD Piano factory programs. The table titles show program IDs and names.

700 New Classical 1

MIDI 6	Disables sympathetic vibrations
MIDI 25	Adds distance
MIDI 67	Soft pedal

The most realistic sound for classical pieces

701 New Classical 2

MIDI 6	Disables sympathetic vibrations
MIDI 25	Adds distance
MIDI 67	Soft pedal

For “big” classical playing

702 Classical Grand3

MIDI 25	Wetter
MIDI 67	Soft pedal

703 DynGrand ClosMic

MIDI 25	Wetter
MIDI 67	Soft pedal

Close-mic dynamic grand; exaggerated filter on softest strikes, for extended dynamic range

704 Jazz Grand

MIDI 6	Disables sympathetic vibrations
MIDI 25	Bigger hall
MIDI 67	Soft pedal

Sounds nice in a jazz combo setting

705 Bright Grand

MIDI 6	Disables sympathetic vibrations
MIDI 22	Treble EQ boost
MIDI 25	Adds distance
MIDI 67	Soft pedal

Sounds like a relatively brightly-voiced piano

706 Songwriter's Pno

MIDI 6	Disables sympathetic vibrations
MIDI 25	Wetter
MIDI 67	Soft pedal

Play simple block chords to accompany a pop or rock song

707 Soft Grand

MIDI 6	Disables sympathetic vibrations
MIDI 25	Wetter

708 Ballad Grand

Mod Wheel	Pitch modulation
MIDI 6	Disables sympathetic vibrations
MIDI 25	Adds chorusing in the reverb

For long sustaining chords in a rock band setting

709 FM & Grand

Mod Wheel	Tremolo
MIDI 6	Disables synth layer
MIDI 22	Timbre Control
MIDI 23	Octave in bass
MIDI 24	Fades frequency-modulation layers
MIDI 25	Wetter
MIDI 26	Chorus wet
MIDI 27	Chorus feedback

710 Hardhammer Piano

MIDI 6	More highs
MIDI 25	Wetter

711 Stage Grand St

MIDI 6	Disables sympathetic vibrations
MIDI 25	Wetter
MIDI 67	Soft pedal

Compressor in studio 749 controls dynamics in live settings

712 Jazz Grand Mono

MIDI 6	Disables sympathetic vibrations
MIDI 25	Reverb time

For use with a mono PA system

713 Rock Grand Mono

MIDI 6	Disables sympathetic vibrations
MIDI 25	Reverb time

For use with a mono PA system

714 GrPno & Strings

Mod Wheel	Lowpass filter on strings
MIDI 6	Strings forward
MIDI 25	Piano delays
MIDI 26	Feedback control (tails longer)

715 GrPno & Pad

Mod Wheel	Fades pad
MIDI 6	Envelope control on pad
MIDI 25	Wetter

716 Grand & Rich Pad

MIDI 26	Feedback control (tails longer)

Velocity controls each note's effects send; accents sound nice sustained

717 Dyn Pno&Pad

MIDI 22	Disables String 1
MIDI 26	Feedback control (tails longer)
MIDI 29	Enables String 2

Velocity controls each note's send to the feedback flangers

718 In Canyon w/Str

MIDI 6	Fades String layer
MIDI 25	Flange wet
MIDI 26	Increases reverb time

719 Mello w/Voxpad

Mod Wheel	Pitch modulation
MIDI 6	Fades string layer
MIDI 25	Flange wet
MIDI 26	Increases reverb time

720 String Chaser

Mod Wheel	Pitch modulation
MIDI 6	Filter
MIDI 25	Adds distance
MIDI 26	Decreases reverb time

721 Bowed Piano

MIDI 6	Fades to bowed layer only
MIDI 25	Controls send to flanger

722 GPno & Puff

MIDI 25	Wetter
MIDI 67	Soft pedal

For percussive playing

723 SynGrand & EPno

Mod Wheel	Manual wah-wah
MIDI 25	Wetter

724 Twang Grand

Mod Wheel	Detune
MIDI 6	Layer balance

725 AlternativePiano

Mod Wheel	Pitch modulation
MIDI 6	Timbre control

726 Affected Grand

Mod Wheel	Filter modulation

727 Robot Grand

MIDI 6	Disables the per-note sample & hold

For staccato repetitive playing (with the arpeggiator, for example)

728 Distorted Grand

MIDI 6	Hipass filter, program-wide

729 Way in the Grand

Mod Wheel	Pitch modulation on layers
MIDI 6	Enables third layer
MIDI 25	Pitch modulation on the reverb
MIDI 26	Drier

Play sparsely!

Appendix G
Vintage Electric Pianos ROM Option

Objects for the Vintage Electric Pianos ROM are included on the SmartMedia card that came with your K2661. The procedure for installing these objects is described in the installation instructions that came with your option kit.

The Vintage Electric Pianos ROM option equips your K2661 with the classic electric piano sounds that are vital to any modern keyboard player. While some electric piano sounds have remained obscure vintage gems, others have attained a level of importance to the keyboardist comparable to that which the Fender Stratocaster or Gibson Les Paul has for the modern guitarist.

Program slots have been assigned and organized for optimum usefulness. The instruments represented in this set are: Fender Rhodes Electric Piano, Wurlitzer Electric Piano, Hohner Pianet, Yamaha CP-80 Electric Grand Piano, and the RMI Electra-Piano.

Each program was created using high-quality audio samples of electric pianos as a starting point. The sounds were then processed using Kurzweil’s V.A.S.T. synthesis engine. This allowed us to apply powerful filters, velocity layers and cross-fades, envelopes and a host of other sound-sculpting tools. In some cases, samples from the K2661’s base-ROM were also used in combination with the Vintage EPs samples in order to add a certain flavor or to enhance the harmonic content of the sound.

The final stage of sound-shaping was done in KDFX, our massive effects processing engine. KDFX played a crucial role in making this a truly ground-breaking project, providing on-board effects unprecedented in both quantity and quality. Effects pedals, speaker cabinets, and recording techniques have all been faithfully replicated, giving the Vintage EPs programs a level of detail and realism never before achieved in any electric piano emulation.

Using V.A.S.T. and KDFX we were able to replicate the exact sounds from dozens of different live and studio recordings of electric pianos. This is why many of the program names in Vintage EPs are derived from song titles and not electric piano model numbers. Also, we have included a section devoted entirely to non-realistic sounds, “hybrids and synths,” where we transformed the Vintage EPs sound sources into a wide variety of sonic textures, ranging from slight mutations to completely unrecognizable new sounds.

We have included detailed charts that list the controller assignments for each program and setup. If a program has been taken from a specific recording, the artist's name and song title have been provided as well.

On the following pages are brief descriptions of each of the instruments sampled for Vintage EPs.

Program Farm Files

In addition to the 100 programs described on the following pages, we have included a number of other programs in the file EPFARM.K26. You can load these programs into the bank of your choice using the K2661’s Disk Mode.

Slider Assignments and KDFX

Here are a few of the guidelines which were used in determining control and slider assignments for the programs in Vintage EPs. These general rules should make it relatively easy to adjust the most basic program settings when first scrolling and playing through the complete set of sounds. Keep in mind that these are *general rules*, and there will be some exceptions. Refer to the table that begins on page G-5 to view the complete controller assignment information for each program.

Many programs in the Vintage Electric Pianos ROM make extensive use of KDFX effects, and the programs were greatly enhanced by the ability to "chain" multiple effects presets on a single KDFX bus. Thirty-one new KDFX studios are included with Vintage EPs, and most of them use this powerful "effects-chaining" feature.

Since you can import studios from any program in the K2661, you may wish to use some of these new studios with other K2661 programs. To select a different studio for a program, press the **ImpFX** soft button while in program-edit mode, select a "target" program containing the desired effects studio, then press **Import**. This imports the target's studio along with all of the FX Mods of the target program. If it does not seem to work right away, go to the Common page, and select a different KDFX pair (usually A or B) for the program.

Fender Rhodes

Produced from 1965-1986 in a number of variations of the original model, the Fender Rhodes is the most widely recognized and easily identified electric piano sound in popular music. The Rhodes played an important role in defining some of the new styles of music that began to emerge in the mid-sixties and early seventies, mainly jazz-fusion, disco and funk, and was adopted quickly by other already established styles such as R&B, rock, pop, blues, and jazz. The Rhodes sound remains popular today and it can be found in a variety of settings: played live by blues, funk and jam bands among others, and on recordings of hip-hop, pop, acid-jazz, and electronica.

Like most other electric pianos, the Rhodes produced its sound electromechanically, with a hammer mechanism striking metal bars or "tines." A damper pedal, much like the one found on an acoustic piano, provided sustain.

The two main models (of which there were a few variations) were the Suitcase Piano and the Stage Piano. The suitcase model was introduced first. Featuring an enclosure equipped with an amplifier, speaker cabinet, and tremolo circuit, the original suitcase model used hammers with felt tips, which were later replaced by ones with neoprene (hard synthetic rubber) tips. The felt hammers gave the early suitcase models a slightly less-bright attack sound and a less-tight decay for each note. The Stage Piano (which did not have an amplifier nor speaker cabinet) was introduced in the early seventies as a more portable alternative to the Suitcase model. Both models were replaced in 1979 by corresponding "Mark II" versions.

Wurlitzer

Similar in both its design and sound to the Rhodes, the Wurlitzer electric piano was nearly as popular, and actually pre-dated the first Rhodes suitcase model. Two basic models were produced, of which there were a few versions: the 100 series, manufactured from c.1954-1967, and the 200 series, which continued from c.1967-1980. Most recordings of the Wurlitzer feature one of the 200 series models.

Using a piano-type action with felt-tipped hammers to strike metal elements, called "reeds", the Wurlitzer employed a damper pedal mechanism to provide sustain, much the way the Rhodes did. The Wurlitzer was smaller than the Rhodes (most had 64 notes), and was available exclusively as a console, which sat on four metal legs. Most models came equipped with at least two built-in speakers, and a tremolo circuit. The Wurlitzer produced a sound which had a slightly more narrow frequency range than the Rhodes; it was "thinner" sounding, blending more easily with other instruments in a mix.

Hohner Pianet

Although featured prominently in a number of classic rock songs by The Beatles, The Zombies and others, the Hohner Pianet remains a lesser-known instrument, its sound having often been mistakenly attributed to the Wurlitzer.

Produced from c.1962-1980 in various console models, the Pianet series differed from the Rhodes and Wurlitzer in a few important areas. Sound was produced by metal reeds, which were plucked by a set of adhesive pads. Also distinguishing the Pianet was the absence of a sustain pedal.

Hohner also manufactured the much more rare Electra-Piano (not to be confused with the RMI Electra-Piano) which featured an enclosure resembling that of an upright piano. Reported to have a hammer mechanism very much like the one found in the Rhodes, the Hohner Electra-Piano featured a built-in amp and four speakers. Led Zeppelin made this sound famous, employing it in a number of hits. Using our Pianet samples and bit of processing, we were able to craft some realistic imitations of the Hohner Electra-Piano, and we've included them in the Pianet section of Vintage EPs.

Yamaha CP-80

Known commonly as the “electric grand”, the CP-80 (88 notes), along with its smaller counterpart, the CP-70 (76 notes), was the product of clever engineering combined with traditional piano-making craftsmanship. Inside the CP-80, are the basic workings of a real acoustic piano, which have been altered to fit into a smaller enclosure. On the outside, the CP-80 looks like a “grand” version of the Rhodes, covered in tolex, with the top portion extending in the rear to accommodate the piano harp inside. Up until the mid-1980’s, when sampled pianos became available, the CP-70/80 was the only instrument capable of providing a decent substitute for a real piano. While it served this purpose well, the CP-70/80 had some unique features, which allowed it to have its own very distinct sound when desired.

With single strings on the lower notes, and double strings on the rest, the CP-70/80 included a modified Yamaha grand piano action and employed piezo-electric transducers in lieu of pickups. The original CP series featured bass and treble tone controls and a tremolo circuit. Later, seven bands of EQ, balanced outputs and MIDI capabilities were added (CP-70/80B and M models). Often used with chorus and compression effects, the CP-80 was known for having more “punch” than an acoustic piano. Production began in 1977 and ended in 1987.

RMI Electra-Piano

Built by Rocky Mount Instruments, a division of the Allen Organ Company, from 1967-1980, the RMI Electra-Piano is the one instrument represented in Vintage EPs which did not produce sound by electromechanical means. With an electronic tone-generator for each note, un-weighted plastic keys, which were not touch-sensitive, and a set of “stops” for sound selection, the RMI more closely resembled an organ than anything else. Both sustain and volume pedals were included with the unit. Most were black tolex-covered consoles with 61 (later 68) keys, and rested on a set of metal legs.

There were five stops on the RMI for tone: Piano, Piano PP, Harpsi, Harpsi PP, and Lute. There were two additional stops; Accenter, which added in an attack “thump” as well as Organ Mode, which extended the decay of held notes.

Though not capable of producing a realistic piano sound, the RMI, with its Harpsi stops did provide a viable “electric harpsichord/clavichord” tone. In addition, the overall sound of the RMI was warm, yet thin and manageable, and lent itself nicely to the use of effects processors and pedals. The RMI was most widely used by progressive-rock bands like Genesis and Yes in the early to mid-1970s, although it has also appeared in a broad variety of other settings.

Vintage EP Programs

ID	Name	Control	Function
600	Model This! Rhds	MWheel	Tremolo Depth
		Data	Tremolo Speed
		MIDI 23	Bass EQ
		MIDI 24	Treble EQ
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Xcouple (tone)
		MIDI 29 (Sw2)	MIDI29+Mwheel gives classic “hard pan” tremolo shape
		Rhodes sound inspired by Chick Corea's album, “Friends”, Peter Gabriel's “Humdrum”, and Kate Bush's “Blow Away”.	
601	Shinin' Xfade	MWheel	Stereo Trem on (KDFX FX1c Simple Panner)
		Data	Trem Rate (for both Trems)
		MIDI 22	Mono Trem on
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Enable Compressor (Comp Ratio)
		Rhds W/Compression & Distortion, recreating the EP on Earth Wind & Fire's “Shinin' Star”.	
602	CleanRhdsEchPlx	MWheel	Stereo Tremolo Depth
		Data	Tremolo Rate
		MIDI22	EQ Brightness
		MIDI 23	Echo Tap2 (Enables 1/8th Note Echoes)
		MIDI 24	Echo Pitch Adjust
		MIDI 25	Echo Wet/Dry
		MIDI 26	Reverb Wet/Dry
		MIDI 27	Echo Mid EQ Gain
		MIDI 28	Echo Bass EQ Gain
		MIDI 29 (Sw2)	Multiple Functions: Changes Reverb Length, Enables Smooth (sine-shaped) tremolo.MIDI 29(Sw2) + Modwheel enables “hard pan” (square shaped) tremolo.
		The sound itself recreates the Rhds on Stevie Wonder's “Living for the City”. Featured in KDFX is Kurzweil's emulation of the classic Echoplex tape delay, with sliders assigned to control the tone and pitch of the echo effect.	

ID	Name	Control	Function
603	AgedTolexPhasSw2	MWheel	Enable Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Phaser Rate
		MIDI 24	Phaser Center Freq (Tone)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Bass EQ (KDFX)
		MIDI 27	Treble EQ (KDFX)
		MIDI 28	Phaser Depth
		MIDI 29 (Sw2)	Enables Phaser
		In “dry” mode, this program emulates the EP sound used by Herbie Hancock on Joni Mitchell's “Mingus” album.	
604	Real 70's Chorus	MWheel	Enables Stereo Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness Control (Filter Cutoff Freq in VAST)
		MIDI 23	Chorus Hi Freq Dampening
		MIDI 24	Reverb Time
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Chorus Depth
		MIDI 27	Chorus Wet/Dry
		MIDI 28	Compression Amount (Lowers Thresh, Increases Ratio)
		MIDI 29 (Sw2)	Lo Freq Cut
605	Studio ClassicEP	MWheel	Enables Stereo Tremolo
		Data	Tremolo Rate
		MIDI 22	Phaser Rate
		MIDI 23	Reverb Hi Freq Dampening (Brightness)
		MIDI 24	PhaserWet/Dry
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Reverb Density
		MIDI 29 (Sw2)	Lo Freq Cut

ID	Name	Control	Function
606	SweetLoretta Amp	MWheel	Enables mono tremolo
		Data	tremolo Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Env release decay
		Emulates Billy Preston's EP sound from the Beatles song, “Get Back”. MIDI28 (H slider) will tighten up the release.(Amount of decay after a note is struck and released)	
607	TheNightFly	MWheel	Phaser Notch Depth (Tone)
		Data	Chorus Rate
		MIDI 22	Phaser Rate
		MIDI 23	Phaser Depth
		MIDI 24	Phaser Center Freq (Tone)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Chorus Hi Freq Dampening
		MIDI 27	Chorus Depth
		MIDI 28	Release Tine Noise Volume
		MIDI 29 (Sw2)	Chorus On/Off
		Recreates Donald Fagen's EP sound on “Green Flower Street” from his album “The NightFly”.	
608	Sugdaddy Mtron	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 23	Mutron (Env Filter) Wet/Dry
		MIDI 25	Delay Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 29 (Sw2)	Enables Flange Effect
		This KDFX studio reproduces the Mutron sound used by Bootsy Collins and Bernie Worrell from Parliament/Funkadelic.	

ID	Name	Control	Function
609	Herbie'sEPWahSw2	MWheel	Enables smooth stereo tremolo
		Data	Tremolo Rate
		MIDI 22	Echo Feedback
		MIDI 23	Behaves exactly like a Wah Pedal when MIDI29 is pressed. (For those without the ccpedal)
		MIDI 24	Echo delay time (for TAP1)
		MIDI 25	Echo Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Feedback Control
		MIDI 29 (Sw2)	Enables Wah Pedal
		CCPedal1	Behaves exactly like a real Wah Pedal when MIDI29 is pressed.
		This program recreates the sound and FX chain used by Herbie Hancock on his albums, “Headhunters” and “Manchild”, with distortion and stereo trem with an FX loop to Wah and Echoplex. MIDI29 button clicks on the wah, and a continuous controller pedal (volume pedal) acts like the wah pedal. Plug the pedal into cc1 jack on the back. MIDI23 (C slider) will also function as the wah for those without the ccpedal. This program (with the wah turned off) also resembles the EP used by Patrick Moraz on Yes's “Relayer” album.	
610	Adjstbl ChDlyRvb	MWheel	Enables stereo tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness Control (Filter cutoff Freq in VAST)
		MIDI 23	Chorus Rate Control
		MIDI 24	Chorus Depth Control
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Delay wet/Dry
		MIDI 27	Chorus feedback Control
		MIDI 28	Chorus pitch Env Shape - Trapezoid or Triangle
		MIDI29(Sw2)	Disables FX

ID	Name	Control	Function
611	XTineRhds RvsRvb	MWheel	Stereo Tremolo Depth
		Data	Tremolo Rate
		MIDI25	Reverse Reverb Wet/Dry
		MIDI26	Reverb Hi Freq Dampening (Brightness)
		MIDI27	Reverb Feedback (Number of Repetitions)
		MIDI28	Reverb Length (Length of Each Rep)
		MIDI29(Sw2)	Enable Smooth Tremolo. MIDI29+Mwheel enables classic "hard pan" tremolo.
		A brighter version of the classic Rhds sound, with extra tine sizzle. Featured in KDFX is a powerful reverse reverb with adjustable parameters assigned to sliders.	
612	BellToneDist RDS	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 24	Static Phaser Notch (Tone)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 29 (Sw2)	MIDI29 alone enables smooth stereo tremolo. MIDI29+MWheel enables classic "hard pan" stereo tremolo.
		Inspired by the great fusion players of the late sixties and early seventies- Jan Hammer of Mahavishnu, Chick Corea w/Return to Forever, and Herbie Hancock w/Miles Davis.	
613	The Phase I'm In	MWheel	Enables Stereo Tremolo
		Data	Tremolo Rate
		MIDI 22	Phaser Rate
		MIDI 23	Phaser Center Freq (Tone)
		MIDI 25	Reverb Wet/Dry
		Inspired by Richard Manuel's phased EP sound on The Band's "The Shape I'm In"	
614	HardStr CompRhds	MWheel	Enables Stereo Tremolo
		Data	Tremolo Rate
		MIDI 22	Lowpass (KDFX EQ) (Brightness)
		MIDI 24	Reverb Time
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Decreases Compression Amount
		MIDI 29 (Sw2)	Changes Reverb Type from Chamber to Plate

ID	Name	Control	Function
615	SizzleTine PhsCh	MWheel	Phaser Notch Depth (Tone)
		Data	Phaser Rate
		MIDI 22	Chorus rate
		MIDI 23	Phaser Depth
		MIDI 24	Phaser Center Freq (Tone)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Chorus Depth
		MIDI 27	Reverb Lowpass (Brightness)
		MIDI 28	Tine Sizzle Amount
		MIDI 29 (Sw2)	Chorus Disable
616	EarlyFusionDstEP	MWheel	Enables Stereo/Mono Tremolo (KDFX Preset)
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Bass EQ (KDFX)
		MIDI 24	Reverb Hi Freq Dampening (Brightness)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Static Phaser Setting (Tone)
		MIDI 27	Distortion Drive and Warmth
		MIDI 28	Env Release Decay
		MIDI 29 (Sw2)	Selects Stereo or Mono Trem
		Inspired by the great fusion players of the late sixties and early seventies- Jan Hammer of Mahavishnu, Chick Corea w/Return to Forever, and Herbie Hancock with Miles Davis. MIDI28 (H Slider) will tighten up the note releases.	
617	Growlin'Electric	MWheel	Enables Mono Tremolo
		Data	Tremolo
		MIDI 22	Filter Resonance (VAST) (Tone)
		MIDI 23	Reverb Hi Freq Dampening (Brightness)
		MIDI 24	Reverb Time/Size
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth and Lowpass (Brightness)
		MIDI 29 (Sw2)	Sample Alt Start (Mellows the Attack)

ID	Name	Control	Function
618	Serious EPno	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Bass EQ (KDFX)
		MIDI 24	Reverb Time
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Reverb Hi Freq Dampening (Brightness)
619	70's HrdStr EP	MWheel	Enables mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Filter Cutoff Freq (VAST) (Brightness)
		MIDI 23	Phaser Center Freq (Tone)
		MIDI 24	Reverb Time/Size
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Phaser Rate
		MIDI 29 (Sw2)	Disables Phaser
620	Rhds No Bell	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Chorus Rate
		MIDI 29 (Sw2)	Lo Freq Cut
621	Hard E Piano	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 25	Reverb Wet/Dry
		This is a template for a creating new sounds.	
622	Soft E Piano	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Filter Cutoff Freq (Brightness)
		MIDI 23	Enables Alt Start (Mellows the Attack)
		MIDI 25	Delay (LaserVerb) wet/Dry
623	Barking Tines	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Turns down/off Thump/Tine Noise
		MIDI29(Sw2)	Switches to Stereo Tremolo

ID	Name	Control	Function
624	Triple Tines	MWheel	Enables mono Tremolo
		Data	Tremolo Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Chorus Mix
		MIDI 27	Delay Mix
		MIDI 29 (Sw2)	(fx3) Chor/Delay On/Off
625	Stay With Me	MWheel	Enables mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Static Phaser Xcouple (Tone)
		This heavily EQed program recreates Ian McLagan's EP sound on the Faces' "Stay With Me."	
626	What'dISay Wurly	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Drive
		MIDI 27	Distortion Warmth
		MIDI 28	Reduces/Turns off thump
		MIDI 29 (Sw2)	Mid EQ Boost (KDFX)
		This is a replica of the classic old Wurly sound used by Ray Charles on "What'd I Say".	
627	RetroVerb Wurly	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Reverb Size Scale (Smaller)
		MIDI 24	Absorption (Reduces perceived reverb size)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Static Phaser Wet/Dry (Tone)

ID	Name	Control	Function
628	VANCradleWillROK	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Turns Down/Off clunks and thumps
		MIDI 29 (Sw2)	Static Phaser Notch (Tone)
		Inspired by Van Halen Wurlitzer into guitar amp sound on “The Cradle Will Rock.”	
629	Supertramp Wrly	MWheel	Enables mono Tremolo
		Data	Tremolo Rate
		MIDI 24	Disable Detuning (VAST)
		MIDI 25	Reverb Wet/Dry
		MIDI 28	Delay Wet/Dry
		MIDI 29 (Sw2)	Disables Phaser
		Rick Davies' s processed Wurly sound with Supertramp inspired this program.	
630	PinkFloydzTheWah	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Behaves like Wah Pedal for those without the CCpedal
		MIDI 24	Reverb Hi Freq Dampening (Brightness)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Static Phaser Xcouple (Tone)
		MIDI 29 (Sw2)	Disables Wah
		CCPedal1	Wah Pedal
		Recreates Pink Floyd’s Wurly w/Wah Pedal sound on “Money” from Dark Side of the Moon.	
631	EQ Vintage Wurly	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Phaser Rate
		MIDI 24	Low Freq Boost (KDFX 1b Res Filter)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Phaser Xcouple (Tone)
		MIDI 29 (Sw2)	Enables Phaser

ID	Name	Control	Function
632	StandnOnTheVerge	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 25	Delay Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 29 (Sw2)	Flange Boost
		Emulates Mutron Wurly on Funkadelic's “Standing on the Verge of Getting' It On”.	
633	WoodstockClunker	MWheel	Disables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Static Phaser Freq (Tone)
		MIDI 23	Static Phaser Out Notch (Tone)
		MIDI 24	Static Phaser FB Notch (Tone)
		MIDI 25	Reverb wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Reduces/Turns off volume for clunks and thumps
		MIDI 29 (Sw2)	Switches Tremolo to Stereo
		Recreates Joni Mitchell's Wurly sound on her version of “Woodstock”. The thumps and clunks (prominent in the recording) can be turned down or off with MIDI28 (H Slider).	
634	Wurly + Alien FX	MWheel	Enables Martian Leslie (Flanger)
		MIDI 22	EQ Mid Boost (VAST)
		MIDI 23	LaserVerb Feedback
		MIDI 24	LaserVerb Delay Length
		MIDI 25	LaserVerb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Turns Down/Off thumps
		MIDI 29 (Sw2)	Enables Martian AutoWah (KDFX 2c EQ Morpher)
		Classic Wurly with distortion, featuring some interesting effects unique to KDFX.	
635	MeanPhaseWurly	Data	Phaser Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 27	Distortion Drive
		MIDI 28	Turns Down/Off Thumps and Clunks
		MIDI 29 (Sw2)	Lo Freq Cut (KDFX EQ)

ID	Name	Control	Function
636	UpcloseHeavyWrly	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Bass EQ (KDFX)
		MIDI 24	Rev Pre Delay Time
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Reverb Decay Time
		MIDI 27	Reverb Room Size
		MIDI 28	Reverb Hi Freq Dampening (Brightness)
637	Lesslie Wurlie	Mwheel	Leslie Fast/Slow
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Vibrato/Chorus Select (KDFX FX2)
		MIDI 24	Flange Rate (Must be enabled)
		MIDI 25	Reverb Wet/Dry
		MIDI 28	Delay Wet/Dry (Must be enabled)
		MIDI 29 (Sw2)	Disables Leslie, Enables Flange/ Delay
638	Soft Wurly	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Enables Alt Start (mellows the attack)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Chorus Wet/Dry
		MIDI 27	Chorus Feedback
		MIDI 28	Reverb Hi Freq Dampening (Brightness)
		MIDI 29 (Sw2)	Enables “Impact”-exaggerated attack

ID	Name	Control	Function
639	Joy to the Piant	MWheel	Enables Tremolo and Vibrato (VAST)
		Data	Tremolo/Vibrato Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Phaser Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Phaser Xcouple (Tone)
		MIDI 29 (Sw2)	Enables Phaser
		Imitates two classic sounds used by Jimmy Greenspoon of Three Dog Night; normally, it produces the EP sound from “Joy to the World”. Moving the mod wheel up will switch to the warbly sound from “Mama Told Me Not to Come”.	
640	TheseEyes	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Phaser Rate
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 29 (Sw2)	Enables Phaser
		Burton Cummings' EP part on The Guess Who's song, “These Eyes” inspired this.	
641	No Quarter	Data	Phaser Rate
		MIDI 22	Brightness (KDFX 2c Resonant Filter)
		MIDI 23	Phaser Center Freq (Tone)
		MIDI 25	Reverb Wet/Dry
		MIDI 27	Distortion Drive
		Recreates John Paul Jones's “bubbly” EP from Led Zeppelin's “No Quarter”.	
642	Black Friday	Data	Phaser Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Phaser Center Freq (Tone)
		MIDI 25	Reverb Wet/Dry
643	StrwyToEPHeaven	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 23	Reverb Hi Freq Dampening (Brightness)
		MIDI 24	Reverb Size
		MIDI 25	Reverb wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		Recreates John Paul Jones' EP sound from Led Zeppelin's “Stairway to Heaven”.	

ID	Name	Control	Function
644	MetalBuzz Piant	MWheel	Enables mono Tremolo
		Data	Tremolo/Vibrato rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST) AND KDFX EQ
		MIDI 23	Tap2 Level (1/8th note echoes)
		MIDI 24	Echoplex Bass Gain
		MIDI 25	Echoplex Wet/Dry
		MIDI 26	Reverb wet/Dry
		MIDI 27	Echoplex Treble gain
		MIDI 28	Turns Down/Off Thumps
		MIDI 29(Sw2)	Enables Vibrato (Pitch in VAST)
645	ThisIsTheWalrus!	MWheel	Increases warble effect (VAST)
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Turns Down/Off thump
		Recreates the distorted Pianet-through-a-Leslie sound used by the Beatles on “I Am The Walrus”.	
646	MistyMountain EP	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Static Phaser Xcouple (Tone)
		Replicates John Paul Jones's EP sound from Led Zeppelin's song, “Misty Mountain Hop”.	
647	Classic Zombies	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Bass EQ (KDFX)
		MIDI 24	Reverb Hi Freq Dampening (Brightness)
		MIDI 25	Reverb Wet/Dry
		Emulates Rod Argent's famous EP sound from the Zombies' classic, “She's Not There.”	

ID	Name	Control	Function
648	Tony's FuzzBox	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 23	Enables Alt Start (Mellows the attack)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Mid EQ Boost
		Inspired by Tony Banks' distorted solo in the Genesis song “The Musical Box”	
649	Queen's Friend	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Gain
		MIDI 28	Static Phaser Xcouple (Tone)
		MIDI 29 (Sw2)	Lo Freq Cut (KDFX EQ)
		This one is modeled after the EP sound on Queen's song “Best Friend”.	
650	Bernie's CP*Funk	MIDI 22	Hi EQ (KDFX 1c)
		MIDI 24	Reverb Size
		MIDI 25	Reverb Wet/Dry
		MIDI 27	Chorus Wet/Dry
		MIDI 28	Compression Ratio
		MIDI 29 (Sw2)	Lo Freq Cut (KDFX EQ)
		Imitates Bernie Worrell's CP-80 sound from The P-Funk AllStars on “Live at the Beverly Theatre Album.” FX1b simulates the natural reverberation of the CP-80's damper sustain mechanism.	
651	CP80 Wallflower	MIDI 22	Hi EQ Boost (KDFX 1c)
		MIDI 23	
		MIDI 24	Reverb Size
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Reverb Hi Freq Dampening (Brightness)
		MIDI 27	Chorus Wet/Dry
		Imitates the rich chorused CP-80 sound on Peter Gabriel's, “Wallflower” from “Security”. NOTE: FX1b is used to simulate the natural reverberation of the CP-80's damper sustain mechanism.	

ID	Name	Control	Function
652	Porta EPiano	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	LoPass Frequency
		MIDI 23	Alt Start Control
		MIDI 24	EnvCtl: Impact
		MIDI 25	(Aux) Hall Level, Reverb Time
		MIDI 26	Delay Mix
		MIDI 27	Chorus Mix
		MIDI 28	Treble Shelf (Aux Reverb)
		MIDI 29 (Sw2)	Dist Amount
653	VideoKilledtheCP	MIDI 25	Reverb Wet/Dry
		MIDI 26	Chorus Wet/Dry
		MIDI 27	Reverb Size
		MIDI 28	Reverb High Freq Dampening (Brightness)
		Emulates the CP-80 sound used by Geoff Downes on Buggles' “Video Killed the Radio Star.” FX1b simulates the natural reverberation of the CP-80's damper sustain mechanism	
654	Red Rain	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	LoPass Frequency
		MIDI 25	(Aux) Reverb Mix
		MIDI 26	Delay Feedback, Mix
		MIDI 27	Chorus Mix
655	OBLA-D CP80	Data	Chorus Rate
		MIDI 22	Brightness (KDFX EQ)
		MIDI 25	Reverb Wet/Dry
		MIDI 27	Chorus wet/Dry
		MIDI 29 (Sw2)	Lo Freq Boost
		FX1b is used to simulate the natural reverberation of the CP-80's damper sustain mechanism.	
656	Dark Elec Grand	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	LoPass Frequency
		MIDI 23	Alt Start Control
		MIDI 24	EnvCtl: Impact
		MIDI 25	(Aux) Hall Level, Reverb Time
		MIDI 26	LFO Rate (Phaser)
		MIDI 27	Feedback Level (Phaser)
		MIDI 28	Bandwidth (Phaser)

ID	Name	Control	Function
657	CP80 All Purpose	MWheel	Enables Tremolo
		Data	Tremolo Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 27	Chorus Wet/Dry
		MIDI 28	Delay Wet/Dry
		MIDI 29 (Sw2)	Selects mono or Stereo Tremolo
658	Bright Live CP80	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	LoPass Frequency
		MIDI 23	Alt Start Control
		MIDI 24	InEQ: Treble
		MIDI 25	(Aux) Hall Level, Reverb Time
		MIDI 26	Phaser Wet/Dry
659	Groovy Dyn Chor	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	LoPass Frequency
		MIDI 23	Alt Start Control
		MIDI 24	EnvCtl: Impact
		MIDI 25	(Aux) Hall Level, Rvb Decay Time
		MIDI 26	Chorus Mix, (fx 3) Delay Feedback
		MIDI 29 (Sw2)	(fx 3) Delay Enable
660	Leave Me Alone	MIDI 25	Reverb Wet/Dry
		MIDI 26	Reverb Time
		MIDI 27	Chorus Wet/Dry
		MIDI 29 (Sw2)	Lo EQ Boost (KDFX EQ)
		Replicates the chorused CP-80 sound used by Billy Joel on “My Life”. NOTE: FX1b is used to simulate the natural reverberation of the Cp-80's damper sustain mechanism.	
661	Thin Space Piano	MIDI 24	Echo (only) Wet/Dry
		MIDI 25	Echo Flange Wet/Dry
		MIDI 26	Phaser Rate
		MIDI 27	Phaser Wet/Dry
		MIDI 28	Reverb Wet/Dry
		MIDI 29 (Sw2)	Lo EQ Boost (KDFX 1a AM Radio)

ID	Name	Control	Function
662	Chorus'd Hall CP	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	LoPass Frequency
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treble
		MIDI 25	(Aux) Send Level
		MIDI 26	Chorus Mix
		MIDI 27	Hi Freq Damping
663	80'sReflectxn EP	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	LoPass Frequency
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treble, LoPass Freq
		MIDI 25	(Aux) Send Level
		MIDI 26	Chorus Mix
		MIDI 27	(fx 1) Wet/Dry Level, Delay Feedback
		MIDI 28	Delay Mix
664	LiveAmp CP Plate	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	LoPass Frequency
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treble
		MIDI 25	(Aux) Send Level
		MIDI 26	Plate Reverb Time
		MIDI 27	Delay Mix
		MIDI 28	Hi Freq Damping
665	Tight Room forEP	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	LoPass Frequency
		MIDI 23	Resonance Env
		MIDI 25	(Aux) Reverb Time
		MIDI 26	(Aux) Send Level, Comp Threshold
		MIDI 27	Compressor Release Time
		MIDI 29	Compressor In/Out
666	Dr. John's RMI	MWheel	Lo Freq Cut (KDFX EQ)
		MIDI 25	Reverb Wet/Dry
		Recreates both of the RMI sounds used by Dr. John on his hit, “Right Place Wrong Time”. Move the modwheel up to produce the thin sound from the intro of the song.	

ID	Name	Control	Function
667	Lamb's Wool	Data	InEQ: Bass Gain
		MIDI 22	Para EQ Amp
		MIDI 23	EnvCtl: Impact
		MIDI 24	EnvCtl: Release Time
		MIDI 25	(Aux) Send Level, Reverb Time
		MIDI 26	Chorus Mix
		MIDI 29 (Sw2)	Accent Layer
668	RMI Phase Flange	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	Stimulator Frequency
		MIDI 23	Stimulator Drive
		MIDI 24	EnvCtl: Release Time
		MIDI 25	(Aux) Send Level
		MIDI 26	Flange Mix
		MIDI 27	Phaser LFO Rate
		MIDI 28	Delay Feedback
		MIDI 29 (Sw2)	Accent Layer
669	RMI Crunch	MWheel	Vibrato
		Data	Vibrato Rate
		MIDI 22	Tube Bass Tone
		MIDI 23	Tube Mid Tone, Alt Control
		MIDI 24	Tube Treb Tone
		MIDI 25	CDR Aux Level
		MIDI 26	Tube Drive
		MIDI 27	Tube warmth
		MIDI 28	InEQ Treb Gain, Bass Gain
670	Wakeman On Ice!!	MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Static Phaser Xcoupling (Tone)
		MIDI 29 (Sw2)	Adds another layer (Thin and Bright)
		Inspired by the RMI sound that Rick Wakeman used live on the “Yessongs” album.	
671	Rael On Broadway	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	Para EQ Amp
		MIDI 23	Alt. Start Toggle
		MIDI 24	EnvCtl: Release Time
		MIDI 25	(Aux) Send Level
		MIDI 26	Dist Drive
		MIDI 27	Chorus Mix
		MIDI 28	Delay Mix, Feedback
		MIDI 29 (Sw2)	Accent Layer

ID	Name	Control	Function
672	MXR InMy Stomach	MIDI 22	Phaser Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 29(Sw2)	Enables Volume Pedal
		CCPedal1	Volume (When Enabled)
		Recreates the sound of Tony Banks running his RMI through an MXR Phaser on the song, "In The Cage" from the Genesis album "The Lamb Lies Down On Broadway."	
673	Electronica RMI	Data	LFO Sweep Filter Rate
		MIDI 22	LoPass Frequency
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treble
		MIDI 25	(Aux) Send Level: LaserVerb
		MIDI 26	LaserVerb Feedback Level
		MIDI 27	LaserVerb Contour
		MIDI 28	LFO Halt
		MIDI 29 (Sw2)	Accent Layer
674	RMI Switch 2	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	BandPass Frequency
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treble
		MIDI 25	Flange LFO FB (fx2), Chorus Mix (fx3)
		MIDI 26	Flange LFO Rate (fx2), Warmth (fx3)
		MIDI 27	(Aux) Send Level
		MIDI 28	(Aux) Delay Tempo
		MIDI 29 (Sw2)	Toggle: Flange + Dist
675	Crocodile Rock	Mpress	Tremolo Depth
		MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	Shaper Amount
		MIDI 23	Alt. Start Toggle
		MIDI 24	EnvCtl: Release Time
		MIDI 25	(Aux) Send Level
676	RoTo RMI	Data	Alt Control
		MIDI 22	Disables Lyr 1
		MIDI 23	EnvCtl: Attack Time
		MIDI 24	EnvCtl: Release Time
		MIDI 25	(Aux) Send Level
		MIDI 26	Rotary Rate Control
		MIDI 27	Rotary In/Out
		MIDI 29 (Sw2)	Disables Lyr 3
677	FrankenRoadz	MWheel	Enables Mono Tremolo
		Data	Tremolo Rate
		MIDI 24	Switches to Stereo Tremolo
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Turns Down/Off thumps and clunks
		MIDI 29 (Sw2)	Cuts level on Layer 7
		This program may be helpful as a template- it has all of the secondary character sounds of a Rhodes (release, tine sizzle, thumps) without the actual meat of the Rhodes sound. Try importing layers in place of layer 7. MIDI28 (H slider) will get the thump sounds under control.	
678	Trace S&H EP	MWheel	Tremolo
		Data	LoPass Resonance, Tremolo Rate
		MIDI 22	LoPass Freq
		MIDI 23	EnvCtl: Decay
		MIDI 24	EnvCtl: Release
		MIDI 25	Impact, (Aux) Reverb Level
		MIDI 26	Flange Feedback Level
		MIDI 27	Flange LFO Tempo
		MIDI 28	(Aux) Reverb wet/dry
679	Baroque Synth	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	BandPass Freq
		MIDI 23	BandPass Width
		MIDI 24	Alt Control RMI
		MIDI 25	(Aux) Plate Wet/Dry/ Reverb Time
		MIDI 26	(Aux) Plate HiFreq Damping
		MIDI 29 (Sw2)	Alt Start Pianet
680	Skunk Art	MWheel	Tremolo - 1 layer, Vibrato other layer, Para EQ control
		MIDI 25	(Aux) Reverb Wet/Dry
		MIDI 26	In EQ Gain, Dist Drive
		MIDI 27	In EQ Freq
		MIDI 28	Reverb Time

ID	Name	Control	Function
681	SliderEP Synth	MWheel	Vibrato, Vibrato Rate, Flange Mid Freq, Tremolo, Tremolo Rate
		Data	Enables “Feedback” Layer, Crossfades Wurly Layer
		MIDI 22	Distortion Mid Gain, Distortion LoPass 6 Freq
		MIDI 23	Impact, Distortion Drive Wurly Layer
		MIDI 24	EnvCtl: Release Wurly
		MIDI 25	(Aux) FDR Level, (Aux) Reverb Time
		MIDI 26	(Aux) Delay Feedback, Flange Feedback
		MIDI 27	(Aux) Flange Tempo
		MIDI 28	(Aux) Delay Mix
		MIDI 29 (Sw2)	(Aux) Delay Feedback
		Mpress	Vibrato, Vibrato Rate, Amplitude LoPass, Tremolo, Tremolo Rate
682	TalkinRingmod	MIDI 25	Reverb Wet/Dry
		MIDI 26	Ring Mod Wet/Dry
		MIDI 27	Ring Mod Osc Freq (Talking Effect)
683	Flurlitzer	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	LoPass Freq, Pitch Amp Mod Oscillator
		MIDI 25	(Aux) Chorus Slapbacks Level
		MIDI 26	Flange Wet/Dry, Flange LFO Tempo, (Aux) LaserContour
		MIDI 27	(Aux) Chorus mix
		MIDI 28	(Aux) LaserDelay Feedback, (Aux) Out Gain
		MIDI 29 (Sw2)	Flange Feedback Level
684	EPno n'Pad	MWheel	Amplitude ctl of pad lyr
		Data	Tremolo On and Rate Ctl
		MIDI 22	Disables Pad Layers
		MIDI 23	EnvCtl: Release EPno layer-longer
		MIDI 24	EnvCtl: Attack/Decay Pad layers
		MIDI 25	(Aux) Reverb Wet/Dry
		MIDI 26	(Aux) Reverb Time
		MIDI 27	(Aux) HiFreq Damping
		MIDI 29 (Sw2)	Disables EPno Layer
685	Pick Up EP	MWheel	Bandpass Filter Freq, EQ, Dist Drive, Rev W/D
		Data	Hi Freq. Stimulation Drive
		MIDI 22	Notch frequency
		MIDI 23	Alt Control
		MIDI 24	EnvCtl: Release Time
		MIDI 25	Reverb wet/Dry w/ ModWheel
		MIDI 26	(Aux) Xcouple
		MIDI 27	Distortion Warmth
		MIDI 28	Filter Resonance
686	Strummed EPiano	MWheel	Tremolo, Tremolo Rate
		Data	Layer Delay
		MIDI 22	HiFreq Stimulation
		MIDI 23	Alt Control
		MIDI 24	(Aux) wet/dry
		MIDI 25	Flange LFO Tempo+Level, (Aux) Mix Chorus, (Aux) Delay Feedback
		MIDI 26	Flange Feedback, Flange Delay Tempo
		MIDI 27	Flange wet/dry, Flange Chorus Rate
		MIDI 28	(Aux) Chorus Slapback Level
		MIDI 29 (Sw2)	EnvCtl: Release
687	Brittle Comper	MWheel	Vibrato
		Data	Bal/Amp, Shaper amount
		MIDI 22	Pitch Control both layers opposite
		MIDI 23	HiPass Freq, Bal/Amp
		MIDI 25	Dist out gain, Dist drive
		MIDI 26	Resonance Gain, Freq
		MIDI 27	Reverb wet/dry
		MIDI 28	Reverb time
		MIDI 29 (Sw2)	Cabinet Preset Switch
688	Cheese Keys	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	Amplitude Ctl of 8va layer
		MIDI 23	EnvCtl: Decay
		MIDI 24	EnvCtl: Release
		MIDI 25	Flange Delay ctl
		MIDI 26	Phaser LFO rate ctl
		MIDI 27	(Aux) Reverb Wet/Dry
		MIDI 29 (Sw2)	Enables lyrs 4-6/disables 1-3

ID	Name	Control	Function
689	RM-Either	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	Disables High 8va Layer
		MIDI 23	Disables Low 8va Layer
		MIDI 24	EnvCtl: Decay
		MIDI 25	(Aux) Reverb Time
		MIDI 26	Phaser LFO Rate Ctl
		MIDI 27	Phaser LFO Depth Ctl
		MIDI 29 (Sw2)	Disables RMI Layer
690	Xylo-Rhoadz	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	Amplitude of Percussive Layer
		MIDI 23	Pitch Ctl (1 8va) of Percussive Layer
		MIDI 24	Pitch Ctl (1 8va) of Tonal Layer
		MIDI 25	Amplitude of Tonal Layer
		MIDI 26	(Aux) Flange-Dly Hall Wet/Dry
		MIDI 29 (Sw2)	Disables Percussive Layer
691	Bubble Snap	MWheel	Vibrato
		Data	Vibrato Rate - Strings enable
		MIDI 22	LoPass Freq Strings
		MIDI 23	LoPass Freq Pianet
		MIDI 24	Resonance
		MIDI 25	(Aux) Gated Reverb
		MIDI 26	Sweep Filter LFO smooth
		MIDI 27	Sweep Filter Max Freq
		MIDI 28	Delay Feedback Level
		MIDI 29 (Sw2)	Sweep Filter LFO Halt+wet/ dry+ min Freq
692	Cypress Pluck	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	Double Notch Separation
		MIDI 23	InEQ Treb Gain
		MIDI 24	Mix Level, InEQ Bass
		MIDI 25	(Aux) PreDelay Plate Level
		MIDI 26	Flange->Pitcher Out Gain, Mix Flange, (Aux) HiFreq Dampening, (Aux) Reverb Time
		MIDI 27	Mix Pitcher, High Freq Dampening Level
		MIDI 28	Pitcher Pitch, LaserVerb Delay Coarse
		MIDI 29 (Sw2)	LaserVerb (bus) switch
693	Electro Fugue	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	BandPass Ctl
		MIDI 23	Timbre (amp) Ctl
		MIDI 24	Alt Control
		MIDI 25	(Aux) Plate Wet/Dry
		MIDI 26	(FX1) Small Chamber Wet/Dry
		MIDI 29 (Sw2)	Alt Control RMI
694	RMI Organ Keys	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	HiPass Freq
		MIDI 23	Amplitude of High 8va Layer
		MIDI 24	EnvCtl: Attack
		MIDI 25	(Aux) CDR Wet/Dry
		MIDI 26	(Aux) CDR Mix Delay/ Reverb
		MIDI 27	(Aux) CDR Delay Time
		MIDI 29 (Sw2)	Disables Percussive Layer
695	Hurtful	MWheel	Vibrato/Tremolo, Vibrato/Tremolo Rate, InEQ HiPass Freq
		Data	Wrap Level
		MIDI 23	Alt Control
		MIDI 24	EnvCtl: Release
		MIDI 25	LVrb Spacing, Reverb wet/dry
		MIDI 26	LaserVerb Delay, Reverb Contour, Reverb out gain
		MIDI 27	(Aux) Flange->Shaper wet/dry and (Aux) out gain
		MIDI 28	InEQ Bass Gain
		Mpress	Vibrato/Tremolo, Vibrato/Tremolo Rate, InEQ HiPass Freq
696	Multi Grand	Data	LoPass Freq
		MIDI 25	(Aux) Reverb Level
		MIDI 26	(Aux) Reverb Time
		MIDI 27	(Aux) HiFreq Dampening
697	Exploring RMI	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	Amplitude of Layer 2
		MIDI 23	Amplitude of Layer 3
		MIDI 24	Pitch Ctl of Lyr 1 (1 8va)
		MIDI 25	Overall FX Wet/Dry
		MIDI 29 (Sw2)	Enables lyr 4/Disables lyr 1

ID	Name	Control	Function
698	Ambient Swells	MWheel	Tremolo
		Data	Tremolo Rate
		MIDI 22	LoPass Freq, LoPass resonance
		MIDI 23	Alt Control
		MIDI 24	Slow Vibrato
		MIDI 25	(Aux) Hall Level
		MIDI 26	Delay Level
699	Raffa's Revenge	MWheel	Real time LaserVerb control (Delay Time)
		Data	Drawbars
		MIDI 22	Drawbars
		MIDI 23	Drawbars
		MIDI 24	Drawbars
		MIDI 25	Drawbars
		MIDI 26	Drawbars
		MIDI 27	Drawbars
		MIDI 28	Drawbars
		MIDI 29 (Sw2)	Enables Laserverb
		SusPedal	Space Leslie (KDFX 2b Wetlip Flange) Slow>Fast
		Shows off some of the K2661's advanced features: It uses a Pianet as the tonewheel source in KB3 mode. The sliders function as drawbars. An EQ Morph effect (KDFX 2c) is triggered by velocity. Real time control of the LaserVerb effect with the mod wheel for outer space effect.	

Vintage EP Setups

Note: SW 1 (above pitch wheel) turns arpeggiator on/off in all setups.

ID	Name	Control	Function
600	Wurly Wash	MWheel	Tremolo Depth
		Data	Phaser LFO Rate
		MIDI 22	BigStrings/FlangeStrings switch
		MIDI 23	Phaser Frequency
		MIDI 25	Reverb Wet/Dry
		MIDI 27	Distortion Drive
		MIDI 28	Turns Down/Off Thumps and Clunks
		MIDI 29 (Sw2)	Lo Freq Cut (KDFX EQ)
601	Solo Access	MWheel	Enable Mono Tremolo
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Xcouple
		Each instrument can be accessed via the buttons above the sliders.	
602	Classic EP Split	MWheel	Stereo Tremolo Depth
		Data	Tremolo Rate
		MIDI22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	InEQ: Bass
		MIDI 24	InEQ: Treble
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Static Phaser Xcouple (Tone)
		MIDI 29 Sw2	Switches from Square Wave to Sine Wave Tremolo
603	Velocity Sw EPs	MWheel	Enables Tremolo
		Data	Tremolo Rate
		MIDI 24	L/R Phase, Rhodes Resonance
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
604	Ballad Grand	MWheel	Enables CP80 Tremolo
		Data	Tremolo Rate
		MIDI 23	Brightness (Filter Cutoff Freq in VAST)
		MIDI 24	Reverb Time
		MIDI 25	(Aux) Send Level, Reverb Time
		MIDI 26	Delay Mix
		MIDI 27	Chorus Wet/Dry
		MIDI 28	In EQ: Treble Freq & Gain

ID	Name	Control	Function
605	Mello MW/EPnoSw2	MWheel	Sw Mellotron Ens Strings/ SoloStrings/Flutes
		Data	Octave Switch (Mellotron)
		MIDI 22	Brightness (Mellotron)
		MIDI 23	Reverb Hi Freq Dampening (Brightness)
		MIDI 24	Phaser Wet/Dry
		MIDI 25	(Aux) Send Level, Reverb Decay Time
		MIDI 26	Phaser LFO Rate
		MIDI 27	Phaser Bandwidth
		MIDI 28	Delay Mix/Feedback Level
		MIDI 29 (Sw2)	Switches RMI/Rhodes
606	Electro Arps	MIDI 22	Parametric EQ Amp
		MIDI 23	Phaser Frequency
		MIDI 25	(Aux) Reverb Wet/Dry
		MIDI 26	Phaser LFO Rate
		MIDI 27	Arp Duration
		MIDI 28	Arp Tempo
		MIDI 29 (Sw2)	Arpeggiator Latch
607	Splitin' Tines	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	Brightness (Filter Cutoff Freq in VAST)
		MIDI 23	Enable Percussive Attack
		MIDI 24	L/R Phase
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
608	Pianette/Wurly	MWheel	Pianet Tremolo Depth
		Data	Pianet Tremolo Rate
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Static Phaser XCouple
609	Funkadelic Split	MWheel	Tremolo Depth with MIDI 29
		Data	LoPass Filter Frequency (Bass Lead)
		MIDI 22	LoPass2 Res Frequency (Bass Lead)
		MIDI 23	Shaper Amt, VAST Dist Drive (Bass Lead), InEQ: Bass (Rhds)
		MIDI 24	Octave Shift (Bass Lead), InEQ: Treble (Rhodes)
		MIDI 25	Echo Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Feedback Control
		MIDI 29 (Sw2)	MIDI29+Mwheel will produce classic “Hard Pan” Tremolo

ID	Name	Control	Function
610	Shuttered Window	MWheel	Vibrato Depth (Fretless Bass)
		MIDI 22	Hi EQ Boost (KDFX 1c)
		MIDI 23	Flute Timbre, Voices Timbre
		MIDI 24	Flute Chiff Amount, Voices Filter Frequency
		MIDI 25	(Aux) Send Level, Reverb Time
		MIDI 26	Delay Mix
		MIDI 27	Chorus Mix
		MIDI 28	In EQ: Treble Frequency & Gain
611	Down and Dirty	MWheel	Tremolo Depth
		Data	Tremolo Rate
		MIDI 22	Hi Freq Drive Mix
		MIDI 23	Enables Alt Start (Fuzz Pianet)
		MIDI 25	(Aux) Send Level, Reverb Time
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
		MIDI 28	Mid Freq Drive, Mix
612	Vintage Rig	MWheel	Tremolo Depth (Rhodes)
		Data	Toggle: Moogy Bass 1 ^ 2
		MIDI 22	LoPass Filter Adj, EnvCtl:Att (Moogy Bass)
		MIDI 23	EnvCtl: Impact (Moogy Bass) In EQ: Bass Gain
		MIDI 24	Tremolo Rate (Rhodes), InEQ: Treble Gain
		MIDI 25	Reverb Wet/Dry
		MIDI 26	AllPass Filter Freq notch setting
		MIDI 27	Distortion Drive, Warmth
		MIDI 29 (Sw2)	Tremolo L/R Phase
613	Phase Shift	MWheel	Vibrato (OB Pad/Brass), Trem Depth (Rvrs Swells)
		Data	Tremolo Rate (Rvrs Swells)
		MIDI 22	Phaser 1 Rate, InEQ: Bass LFO Rate
		MIDI 23	Toggle: OB Pad ^ OB Brass
		MIDI 25	(Aux) Reverb Wet/Dry
		MIDI 26	Phaser 2 Rate
		MIDI 29 (Sw2)	(Aux) A->B Configuration (Rvb->Cmp)
		CCPedal 1	RMI Organ Volume
614	Con Fusion	MWheel	Phaser Notch Depth (Tone), Bass Vibrato
		Data	Toggle: Warm Bass 1 ^ 2
		MIDI 22	Phaser Rate
		MIDI 23	Phaser Depth
		MIDI 24	Phaser Center Freq (Tone)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Chorus Hi Freq Dampening
		MIDI 27	Chorus Depth
		MIDI 28	Release Tine Noise Volume
		MIDI 29 (Sw2)	Chorus On/Off
615	Ringin' Soundboard	MWheel	Tremolo
		Data	Tremolo Rate, InEQ: Bass Gain
		MIDI 22	Double Notch Separation
		MIDI 23	InEQ Treb Gain
		MIDI 24	Mix Level, InEQ Bass
		MIDI 25	(Aux) Send Level, Reverb Time
		MIDI 26	Chorus Wet/Dry
616	SynBrass/ CPno	MWheel	Vibrato (SynBrass + Strings), Trem Depth (EP)
		Data	LoPass Freq, EnvCtl: Att+Dec, Toggle: MelloStr^ShineOn
		MIDI 22	Enables 5th (Syn Brass), LoPass (Strings)
		MIDI 23	Bass EQ (KDFX)
		MIDI 24	Reverb Hi Freq Dampening (Brightness)
		MIDI 25	(Aux) Send Level, Reverb Time
		MIDI 26	Delay Mix
		MIDI 27	Chorus Mix
		MIDI 28	In EQ: Treble Frequency & Gain
617	Black Sheep	Data	InEQ: Bass Gain
		MIDI 22	Para EQ Amp (RMI)
		MIDI 23	EnvCtl: Impact (RMI)
		MIDI 24	Toggle: Clavs 1^2^3
		MIDI 25	(Aux) Send Level, Reverb Time
		MIDI 26	Chorus Mix
		MIDI 29 (Sw2)	Arp Latch
618	Acous/Elec Split	MWheel	Tremolo Depth (Rhodes)
		Data	Tremolo Rate (Rhodes)
		MIDI 22	Brightness (Filter Cutoff Freq in VAST) (CP80)
		MIDI 25	Reverb Wet/Dry
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive
619	Strings & P'anet	Data	AllPass Frequency (Strings)
		MIDI 22	Filter Cutoff Freq (VAST) (Pianet)
		MIDI 23	Phaser Center Freq (Tone), (Pianet)
		MIDI 26	Distortion Warmth
		MIDI 27	Distortion Drive

Vintage Electric Piano Keymaps

ID	Keymap	ID	Keymap
600	Rhoadz Hard	615	Wurly Key Rel
601	Rhoadz Soft	620	RMI_EP
602	Rhoadz 2-vel	621	RMI_ACC
603	Rhoadz Thump	622	RMI_accenter lo
604	Rhoadz Soft c7	623	RMI_accenter hi
610	Wurly Hard	624	RMI_accenter jk
611	Wurly Med	625	RMI ds6
612	Wurly Soft	630	CP80 E Grand
613	Wurly 3-vel	635	Pianette
614	Wurly Thump	636	Fast Pianette

Vintage Electric Piano Samples

ID	Sample	ID	Sample
600	Rhoadz	617	RMI Accenter
603	Rhoadz Thump	618	RMI_accenter lo
605	Wurly	619	RMI_accenter hi
608	Wurly Thump	620	CP80 E Grand
609	Wurly Key Rel	623	Pianette
615	RMI Electra Piano	624	Fast Pianette

Vintage Electric Piano Studios

ID	Studio	ID	Studio	ID	Studio
600	ChorChorCDR Spac	611	WurlyEQ Amp Rvb1	621	Chorus Studio
601	Walrus Studio	612	Obla-DCP80	622	The Shape I'm In
602	DistCabEP Box	613	Pnt Amp	623	NoQuarter2Phaser
603	DistCabEnh Rm	614	CP80 Wallflower	624	CP80 Wallflower2
604	Phase SmSpace	615	Rhds TrmCab	625	Rhds TrmCab 2
605	SuperROAR!	616	CP80 Simple	626	PhaseCompDist
606	WurlyEQ Amp rvb2	617	PntEQ Amp Rvb	627	RvrsRvbFlgAmp
607	auxEnvSp4T GtVrb	618	Joni's Wurly	628	Tony's MXR PHS90
608	EP Amp	619	RealEchoPlx Rhds	629	Reverse ReverbSB
609	EP AmpEchWah cmb	620	Rhds St Tremolo	630	Barking Amp
610	Flaming Mtron				

Appendix H
General MIDI

General MIDI (GM) is an addition to the original MIDI specification that assigns sounds to specific channel numbers, program numbers, and note values. The K2661's GM Mode feature (described in Chapter 11 of the *Musician's Guide*) sets up your instrument for GM in a single step. Using General MIDI, you can share song files between different devices with reasonably consistent performance.

Many GM song files are commercially available, and they'll sound great on your K2661.

Inside GM Mode

Here's what happens when you enable GM Mode from the Master Page:

- On all channels except channel 10 (which GM uses for drums), you will see only the 128 GM programs. On MIDI channel 10, you will see the eight drum kits.
- The K2661 will modify the following entries in the master table:

    - FX mode (GM uses Master mode)
    - FX channel (GM uses None)
    - FX studio (GM uses the studio selected in GM Studio set on the Master: GM page)
    - Receive velocity map (GM uses the GM Receive Velocity Map)
    - progChgType (GM uses 0-127 mode)

    Old settings will be remembered, however, so that when you turn GM Mode off the K2661 will restore your previous settings.
- Volume and expression controllers are mapped to a special GM curve, as in "GS" synths. (GS is a superset of General MIDI that is used by the Roland Sound Canvas and other products.)
- GM drum kits are mapped across program number space as in the "GS" synths, and have exclusive zones included with them.
- Program changes sent to the K2661 when it is in GM Mode will only select programs from the GM program set.

Setups, Songs, and QA Banks created outside of GM Mode will not point to the correct programs within GM Mode (although you may find the results "interesting"). Similarly, Setups, Songs, and QA Banks created within GM Mode will not point to the correct programs when you leave GM Mode. For this reason, when you create Setups, Songs, or QA Banks within GM Mode you may want to append the letters "GM" to the object's name and/or store the objects only in certain banks.

General MIDI Programs

The table below shows the 128 General MIDI programs. The ID numbers shown are the locations that these programs will occupy in GM Mode. In Standard Mode the program numbers will be 400-527.

You can create your own GM sets as well, provided that you store the programs at 400-527 and the drum kits at 528-535.

1	Grand Piano	33	Acoustic Bass	65	Soprano Sax	97	Ice Rain
2	Bright Piano	34	Fingered Bass	66	Alto Sax	98	Soundtrack
3	Electric Grand	35	Picked Bass	67	Tenor Sax	99	Crystal
4	Honky-Tonk Piano	36	Fretless Bass	68	Baritone Sax	100	Atmosphere
5	Elec Piano 1	37	Slap Bass 1	69	Oboe	101	Brightness
6	Elec Piano 2	38	Slap Bass 2	70	English Horn	102	Goblins
7	Harpsichord	39	Synth Bass 1	71	Bassoon	103	Echo Drops
8	Clavinet	40	Synth Bass 2	72	Clarinet	104	Sci-fi Pad
9	Celeste	41	Violin	73	Piccolo	105	Sitar
10	Glockenspiel	42	Viola	74	Flute	106	Banjo
11	Music Box	43	Cello	75	Recorder	107	Shamisen
12	Vibraphone	44	Contrabass	76	Pan Flute	108	Koto
13	Marimba	45	Tremolo Strings	77	Blown Bottle	109	Kalimba
14	Xylophone	46	Pizzicato String	78	Shakuhachi	110	Bagpipe
15	Tubular Bells	47	Plucked Harp	79	Whistle	111	Fiddle
16	Dulcimer	48	Timpani	80	Ocarina	112	Shanai
17	Drawbar Organ	49	Ensemble Strings	81	Square Wave	113	Tinkle Bell
18	Perc Organ	50	Slow Strings	82	Sawtooth Wave	114	Agogo
19	Rock Organ	51	Synth Strings 1	83	Synth Calliope	115	Steel Drums
20	Church Organ	52	Synth Strings 2	84	Chiff Lead	116	Woodblock
21	Reed Organ	53	Choir Oohs	85	Charang	117	Taiko Drum
22	Accordion	54	Voice Oohs	86	Solo Vox	118	Melodic Toms
23	Harmonica	55	Synth Vox	87	Fifths Saw Wave	119	Synth Drums
24	Bandoneon	56	Orchestra Hit	88	Bass & Lead	120	Reverse Cymbal
25	Nylon Str Guitar	57	Trumpet	89	Fantasia Pad	121	Gtr Fret Noise
26	Steel Str Guitar	58	Trombone	90	Warm Pad	122	Breath Noise
27	Jazz Guitar	59	Tuba	91	Poly Synth Pad	123	Seashore
28	Clean Guitar	60	Muted Trumpet	92	Space Voice Pad	124	Birds
29	Muted Guitar	61	French Horn	93	Bowed Glass Pad	125	Telephone
30	Overdrive Guitar	62	Brass Section	94	Metallic Pad	126	Helicopter
31	Distorted Guitar	63	Synth Brass 1	95	Halo Pad	127	Applause
32	Guitar Harmonics	64	Synth Brass 2	96	Sweep Pad	128	Gun Shot

GM Drum Kits

The table below lists the drum kits provided with GM Mode for the K2661. The location for the kits (as shown in columns 1 and 2 of the table) will depend on whether or not GM Mode is enabled. You can also create own GM drum kits and store them at locations 528-535.

GM Mode Program No.	Standard Mode Program No.	Drum Kit Name
1	528	Standard Kit Pan
9	529	Room Kit Pan
17	530	Power Kit Pan
25	531	Synth Kit Pan
26	532	Analog Kit Pan
33	533	Jazz Kit Pan
41	534	Brush Kit Pan
49	535	Orch Kit Pan

Standard Mode Controller Assignments

ID	Name	Ctrl	Function
400	Grand Piano	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	Aux Lo Pass
		MIDI 25	L/R PreDelay Time
401	Bright Piano	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	Aux Lo Pass
		MIDI 25	L/R PreDelay Time
402	Electric Grand	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	Aux Lo Pass
		MIDI 25	L/R PreDelay Time
403	Honky Tonk	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
404	Elec Piano 1	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
405	Elec Piano 2	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
406	Harpsichord	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
407	Clavinet	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
408	Celeste	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
409	Glockenspiel	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
410	Music Box	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
411	Vibraphone	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
412	Marimba	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
413	Xylophone	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
414	Tubular Bell	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
415	Santur	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
		MIDI 26	Absorption
416	Drawbar Organ	MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 25	Vib/Chor In/Out
		MIDI 26	Aux Level
		MIDI 29 (Sw2)	Leslie Fast/Slow

ID	Name	Ctrl	Function
417	Perc Organ	MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	Vib/Chor In/Out
		MIDI 26	Aux Level
		MIDI 29 (Sw2)	Leslie Fast/Slow
418	Rock Organ	MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	Vib/Chor In/Out
		MIDI 26	Aux Level
		MIDI 29 (Sw2)	Leslie Fast/Slow
419	Church Organ	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
420	Reed Organ	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
421	Accordion	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
		MIDI 26	LFO Rate
422	Harmonica	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
		MIDI 26	LFO Rate
423	Bandoneon	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
424	Nylon Guitar	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
425	Steel Str Guitar	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
426	Jazz Guitar	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
427	Clean Elec Guitar	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
		MIDI 29 (Sw2)	Delay ON/Off
428	Muted Guitar	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
		MIDI 29 (Sw2)	Delay ON/Off
429	OD Guitar	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	Lo Pass
		MIDI 25	L/R PreDelay Time
430	Dist Guitar	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	L/R Delay Fdbk
		MIDI 29 (Sw2)	Alt start
431	Gtr Harmonics	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	L/R Delay Fdbk
432	Acoustic Bass	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
		MIDI 26	FX1 Aux Level
433	Finger Bass	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
434	Pick Bass	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
435	Fretless Bass	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
436	Slap Bass 1	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
437	Slap Bass 2	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
438	Synth Bass 1	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
439	Synth Bass 2	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
440	Violin	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
441	Viola	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
442	Cello	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Absorption
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
443	Contrabass	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
444	Trem Strings	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
445	Pizz Strings	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
446	Harp	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
447	Timpani	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
448	Strings	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
449	Slo Strings	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
450	Syn Strings 1	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
451	Syn Strings 2	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
452	Choir Aahs	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
453	Voice Doos	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
		MIDI 26	FX1 Aux Level
454	Syn Vox	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
455	Orchestra Hit	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
456	Trumpet	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
457	Trombone	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
458	Tuba	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
459	Muted Trumpet	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
460	French Horns	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
461	Brass Section	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
462	Synth Brass 1	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
463	Synth Brass 2	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
464	Soprano Sax	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
465	Alto Sax	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
466	Tenor Sax	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
467	Baritone Sax	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
468	Oboe	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
469	English Horn	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
470	Bassoon	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
471	Clarinet	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
472	Piccolo	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
473	Flute	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
474	Recorder	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
475	Pan Flute	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
476	Bottle Blow	MWheel	Vibrato
		MIDI 22	"Wet/Dry level, Feedback Level"
		MIDI 23	L/R Dly Time
477	Shakuhachi	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
478	Whistle	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
479	Ocarina	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
480	Square Wave	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
481	Saw Wave	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
482	Syn Calliope	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
483	Chiffer Lead	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
484	Charang	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
485	Solo Vox	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
486	5th Saw Wave	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	Mix Delay
487	Bass & Lead	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
488	Fantasia	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
489	Warm Pad	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
490	Poly Synth	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
491	Space Voice	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
492	Bowed Glass	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
493	Metallic Pad	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
494	Halo Pad	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
495	Sweep Pad	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
496	Ice Rain	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	L/R Mix Delay
		MIDI 24	L/R Delay Feedback
		MIDI 25	Delay Tempo
497	Soundtrack	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	L/R Mix Reverb
		MIDI 24	L/R Delay Time
498	Crystal	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
499	Atmosphere	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
500	Brightness	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time/Absorption
		MIDI 24	HF Dampening
		MIDI 25	L/R PreDelay Time
501	Goblins	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	Lo Pass
		MIDI 25	L/R PreDelay Time
		MIDI 26	Aux LateRvb Time
502	Echo Drop	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
503	Star Theme	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
504	Sitar	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
505	Banjo	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	"L/R PreDelay Time, Build Time"

ID	Name	Ctrl	Function
506	Shamisen	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
507	Koto	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
508	Kalimba	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
509	Bagpipe	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
510	Fiddle	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
511	Shanai	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
		MIDI 26	Wet/Dry of Delay
512	Tinkle Bell	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
		MIDI 26	Mix Delay
513	Agogo	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
514	Steel Drum	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
515	Woodblock	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
516	Taiko Drum	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
517	Melodic Drum	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
		MIDI 29 (Sw2)	Aux Lvl
518	Synth Drum	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
519	Rev Cymbal	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
520	Gtr. Fret Noise	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
521	Breath Noise	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
522	Seashore	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
523	Birds	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
524	Telephone	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
525	Helicopter	MWheel	Vibrato
		MIDI 22	Wet/Dry level
526	Applause	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
527	Gunshot	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
528	Standard Kit	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
529	Room Kit	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
530	Power Kit	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
531	Synth Kit	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	LFO Period
		MIDI 24	HF Damping
		MIDI 25	Min/Max Frequency
532	Analog Kit	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	LFO Period
		MIDI 24	HF Damping
		MIDI 25	Min/Max Frequency
533	Jazz Kit	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time

ID	Name	Ctrl	Function
534	Brush Kit	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time
535	Orch Kit	MWheel	Vibrato
		MIDI 22	Reverb Wet/Dry level
		MIDI 23	Reverb Time
		MIDI 24	HF Damping
		MIDI 25	L/R PreDelay Time

Appendix I
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

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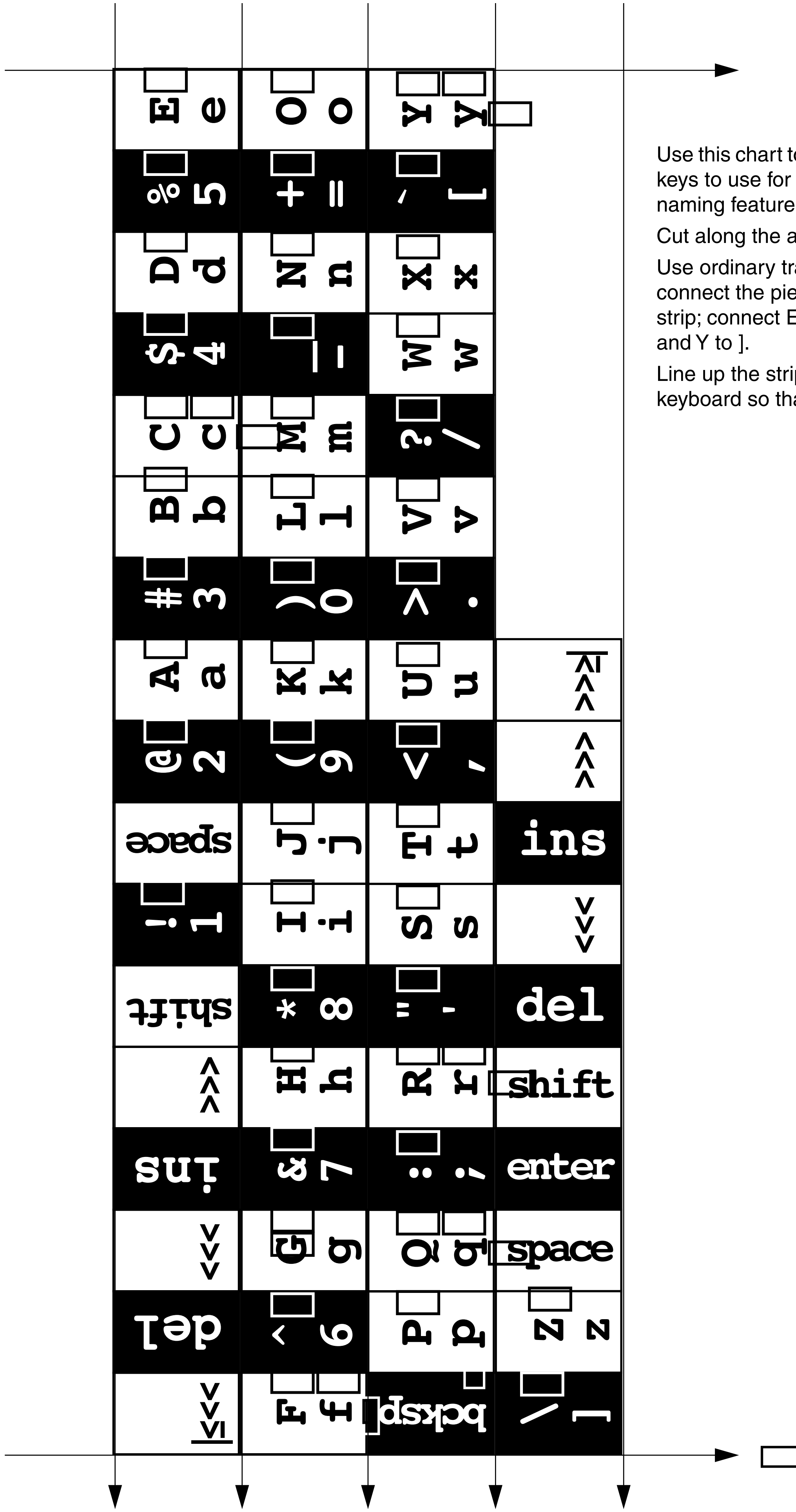

Use this chart to help you learn the keys to use for the keyboard naming feature.

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.