Chapter 15
Advanced KDFX

This chapter describes the organization of KDFX in detail.

Personalities

KDFX can be said to have two distinct "personalities," depending on the setting of the FX Mode parameter on the Effects-mode page.

If FX Mode is set to **Master**, then all KDFX parameters are set within the current studio, and remain unchanged unless you edit the studio or any of its FX presets.

If FX Mode is set to anything else —**Program**, **Setup**, or **Auto**—then one or more parameters within the studio may be under the control of an outside source, such as MIDI or one of the K2600's control sources, and can be continuously changed in real time without editing the studio or any FX presets.

For the sake of clarity, we'll begin by discussing KDFX only in Master mode. For information about real-time control, turn to page 15-21.

*For starters, set the FX Mode parameter on the Effects-mode page to **Master** and the Channel FX Chan parameter to **None** before going further into this chapter.*

Navigating KDFX

The largest component in KDFX is the studio. The studio encompasses all of KDFX's signal routings, processing algorithms, and processing parameters. When you change any parameter in a studio, you are potentially creating a new studio, just as changing a parameter in a program creates a different program (if you save it). The user interface within a studio is organized according to the following diagram:

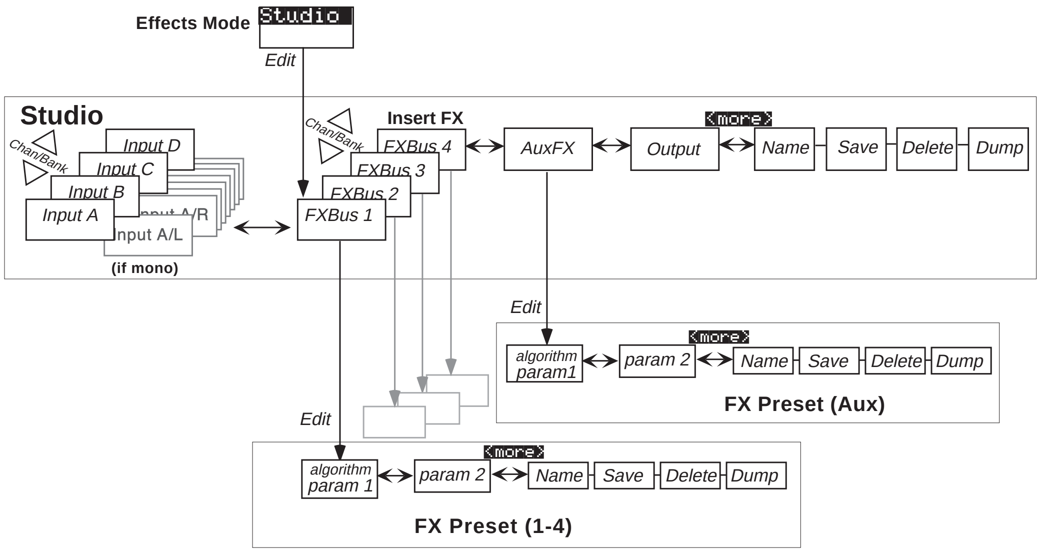


Figure 15-1 The KDFX User Interface

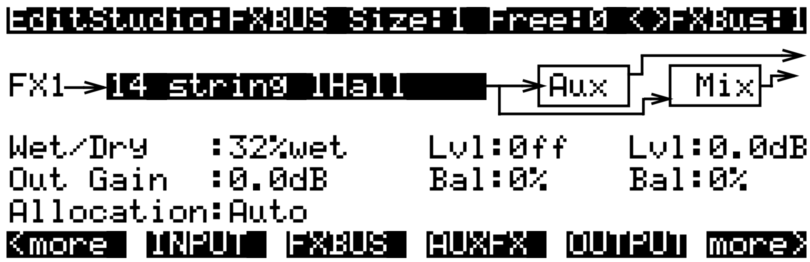
To select a studio, go to the main Effects page if you're not already there (from Program mode, press **Effects** to go into Effects mode), and move the cursor so the studio name is highlighted.

```
EffectMode       XPose:0ST      <>Channel:1
FX Mode:Master
FX Chan:None
Studio :2 HallFlngCdD Room
Effect :1 Sweet Hall
Wet/Dry:0%Wet
Dither :Medium      DigOut :16 Bit
Octav- Octav+ Panic         Chan-  Chan+
```

FXBus

When the studio is highlighted, pressing **Edit** goes to the Studio Editor (EditStudio) level, and the FXBUS page. There are four FXBUS pages, one for each FXBus in the studio. These four buses are the inputs to KDFX, and receive the output from the K2600's sound engine, as defined on the OUTPUT page in the Program Editor.

Select the desired FXBUS page using the **Channel**/**Bank Up** and **Down** buttons. If you have just entered the Studio Editor, the first FXBUS page you see will be the one for FXbus1. Once you are inside the Studio Editor, however, when you press the **FXBUS** soft button from another page, it will take you to whichever FXBUS page you were *last* looking at.

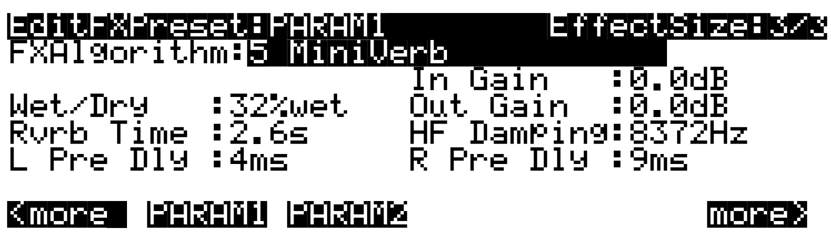


The four FXBuses are the equivalent of four effects processors inserted into the effects loop of a mixing console. Therefore, they are also known as the *Insert effects*.

Parameters

When you are on an FXBUS page, highlighting the name of the FX preset (or any of the parameters directly below it) and pressing **Edit** accesses the first page of parameters for that FX preset. An FX preset is an object within the studio, much like a keymap is an object within a program. The same FX preset can be used in more than one studio, or more than once in the same studio (provided you don't run out of PAUs).

The first EditFXPreset Parameter page includes the algorithm on which the FX preset is based. The soft buttons take you to additional pages of parameters. Depending on how complex the algorithm is, there may be as many as four parameter pages in an FX preset. Algorithms are in the KDFX ROM, and are not normally changeable, deletable, or saveable by the user. Like ROM samples, they are simply always there. (You can load additional algorithms from disk, however, as they become available from Kurzweil.)

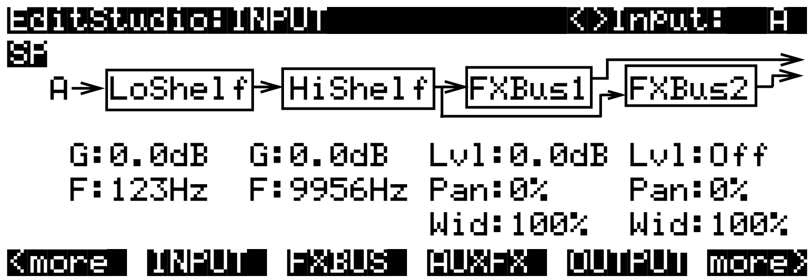


Pressing the **<more>** soft buttons gives you access to the Name, Save, Delete, and Dump pages for the FX preset.

Pressing **Exit** goes back to the FXBUS page, and if you have made any changes in the FX preset you will be prompted to save it. If the FX preset in this bus is the same as on another bus (either in this studio or another), then any changes you make (and save) will affect all buses using that FX preset.

Input

From the FXBUS page, or anywhere inside the Studio Editor, pressing the **INPUT** soft button brings you to the Input A page.



The other Input pages—B, C, and D—are selected by using the **Chan**/**Bank Up** and **Down** buttons. Depending on how the inputs are configured—stereo or mono—there will be from four to eight Input pages.

Similar to the FXBUS pages, the first time you look at an input page after entering the Studio Editor, it will be the Input A page (or, if it's mono, the Input A/L page). Once you are in the Studio Editor, when you press the **INPUT** soft button from another page, it will take you to whichever input page you were *last* looking at.

Aux FX

From inside the Studio Editor, a soft button accesses the AUXFX page. This is a separate effects bus, which can be used by itself, or in a chain following one or more of the FXBuses.

Like the FXBuses, you can view and edit the FX preset's parameters, including its algorithm, by highlighting the FX preset's name and pressing **Edit**. As on the insert FXBuses, the FX preset on the Aux bus has up to four pages of parameters, and the **<more>** soft buttons access Name, Save, Delete, and Dump pages for the FX preset. The same FX preset can be used in the Aux bus as in any of the insert FXBuses.

The AUXFX page can also be accessed from any of the FXBUS pages by placing the cursor on the box labelled Aux and pressing **Edit**.

Pressing **Exit** goes back to the AUXFX page on the EditStudio level. If you have made any changes in the FX preset, you will be prompted to save the FX preset.

Output

In the Studio Editor, pressing the **OUTPUT** soft button accesses the OUTPUT page, where the KDFX's "virtual" outputs are assigned to the K2600's physical outputs.

```
EditStudio:OUTPUT

Mix Lvl:0.0dB        Output A:Mix
Mix Bal:0%           Output B:Off
                     Output C:Off
                     Output D:Off

<more  INPUT  FXBUS  AUXFX  OUTPUT  more>
```

The OUTPUT page can also be accessed from any of the FXBUS pages by placing the cursor on the box labelled Mix and pressing **Edit**.

Name, Save, Delete, Dump

From any of the EditStudio pages, pressing either of the **<more>** soft buttons accesses Name, Save, Delete, and Dump pages for the studio. Studios are stored in RAM, like K2600 programs, and when a studio is recalled, all of its associated FX presets and parameters are recalled with it. Studios in ROM occupy slots in the zeros, 100s, 700s, and 800s banks. You may override these if you like, or use the RAM banks (200-699 and 800-999) for your studios. The ROM studios are always there, and if you delete a studio that you've stored in a ROM slot, the original ROM studio will pop up in its place.

When you save a studio, you can also rename it, using the standard naming dialog:

```
EditStudio:Rename

Studio Name:    HallFlngChD Room

Delete Insert  <<<    >>>    OK   Cancel
```

The Compare and FX Bypass Buttons

As with all K2600 objects, the **Compare** (**Disk**) button lets you switch back and forth between the last saved version of whatever you are editing and its current state.

If you are on a page at the EditStudio level, **Compare** toggles between the last-saved and current versions of the studio. If you are inside an FX preset, on the EditFXPreset level, **Compare** toggles between the last-saved and current versions of the FX preset.

If you have changed any algorithms in an FX preset or studio during the current editing session, the **Compare** button will switch back to the old algorithms. This can create some short-term "holes" in the audio output when the signal momentarily goes dry—see the section on switching studios in real time on page 15-21.

The **Effects**/**FX Bypass** button, when you are in the Studio Editor, bypasses all of the FX presets (all of the Insert FX and the Aux FX) in the current studio, so that you can hear the signals without processing. It does not, however, change the EQs, gains and balances, or signal routings—those will continue to affect the signal you hear.

Exploring the Studio Parameters

We'll explore the parameters within the studio in the order in which they affect the signal path, starting on the Input page.

Input Section

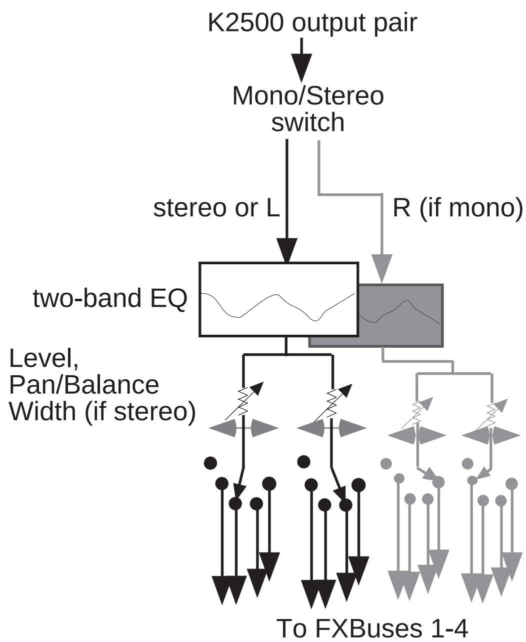


Figure 15-2 The KDFX Input Editor

Inputs are referred to as A, B, C, and D, and correspond to the four output buses (KDFX-A through KDFX-D) from the K2600's Program Editor. These signals can be treated as stereo pairs or as individual mono signals. This is determined by the first parameter on an Input page, the Mono/Stereo switch.

If this switch is set to **M**, then the left and right channels of the selected program output pair are split up, and each is given its own Input page, with EQ and FXBus routings. If it is set to **SP** (Stereo with Pan) or **SB** (Stereo with Balance), the two channels of the pair are processed in parallel.

Selecting the Mono/Stereo mode on one input bus does not affect any of the others, and you can have any combination of stereo and mono inputs in a studio. Therefore, there can be anywhere from four to eight Input pages in a studio.

The **Chan**/**Bank Up** and **Down** buttons let you move among the Input pages.

The Arrow Meter

On an Input page, whenever there is signal present on its bus, the arrow next to the letter of the bus flashes. This is a good way to check that you have set up your program output routings correctly. More on this later.

EQ

The input signal first passes through two equalizers. These equalizers are independent of each other, but the signal is chained: it goes through the left one, then the right one. Each equalizer has a frequency (F) control and, depending on the mode, a gain (G) control. The mode of each EQ module is changed by placing the cursor in the appropriate block and turning the Alpha wheel or using the **Plus/Minus** buttons. There are eight modes for the first EQ, and six for the second:

None No effect, the signal passes through unchanged.

LoShelf Boosts or cuts frequencies below the F value by G decibels.

HiShelf Boosts or cuts frequencies above the F value by G decibels.

LoPass1 Cuts frequencies above the F level with a 6 dB/oct (1-pole) slope.

LoPass2 Cuts frequencies above the F level with a 12 dB/oct (2-pole) slope.

HiPass1 Cuts frequencies below the F level with a 6 dB/oct (1-pole) slope.

HiPass2 Not available on the second EQ. Cuts frequencies below the F level with a steeper 12 db/oct (2-pole) slope.

ParaMid Not available on the second EQ. Provides a cut or boost centered around the F frequency. The bandwidth of the equalizer is two octaves. There is an illustration of the action of this equalizer mode on page 16-28.

FXBus Sends

Following the equalizers are the insert FXBus sends. Each input has two sends. Change the destination of each send by placing the cursor on it and doing the usual thing with the Alpha wheel or **Plus/Minus** buttons.

Either send can be assigned to any of the four FXBuses, or to **None**, with one exception: the two sends on a particular input cannot both be assigned to the same FXBus. So, for example, if the first send on Input B/L is assigned to FXBus2, the second send from Input B/L cannot also be assigned to FXBus2. You can, however, assign as many *different* inputs to the same FXBus as you like—including the two channels from a mono pair.

Each of the FXBus sends has a level parameter (Lvl) that determines the gain of the signal going to that send. Maximum level is **24.0 dB**, and minimum is **-79.0 dB**—there is also an **Off** position. **0.0 dB** is unity gain.

The FXBus sends are stereo, and if an input is stereo, both channels go to the send.

Mono Inputs (M)

If an input is mono, then each of its FXBus sends has a Pan parameter. This determines how the signal is distributed between the left and right channels going to the FXBus: **-100%** is left channel only, **100%** is right channel only, and **0%** is both channels equally.

Stereo Inputs with Pan (SP)

If the input is set to **SP**, then each FXBus send has a Pan parameter and a Width parameter. The Width parameter determines how much separation there will be between the left and right input signals as they are sent to the FXBus: assuming Pan is set to **0%**, a Width of **100%** means the signals will be completely separate, while **0%** means they will be combined into “dual mono.” Negative numbers flip the channels around, so that **-100%** means the channels are separate, but with left and right reversed, while **-50%** means they are reversed and partially blended.

The Pan control maintains the stereo image, but “tilts” it one direction or the other. At **0%** there is no change to the signal, while at **100%** it all goes to the right channel. At **50%**, what had been hard left will now be in the center, and what had been in the center will now be halfway between center and right. Negative values tilt the signal to the left.

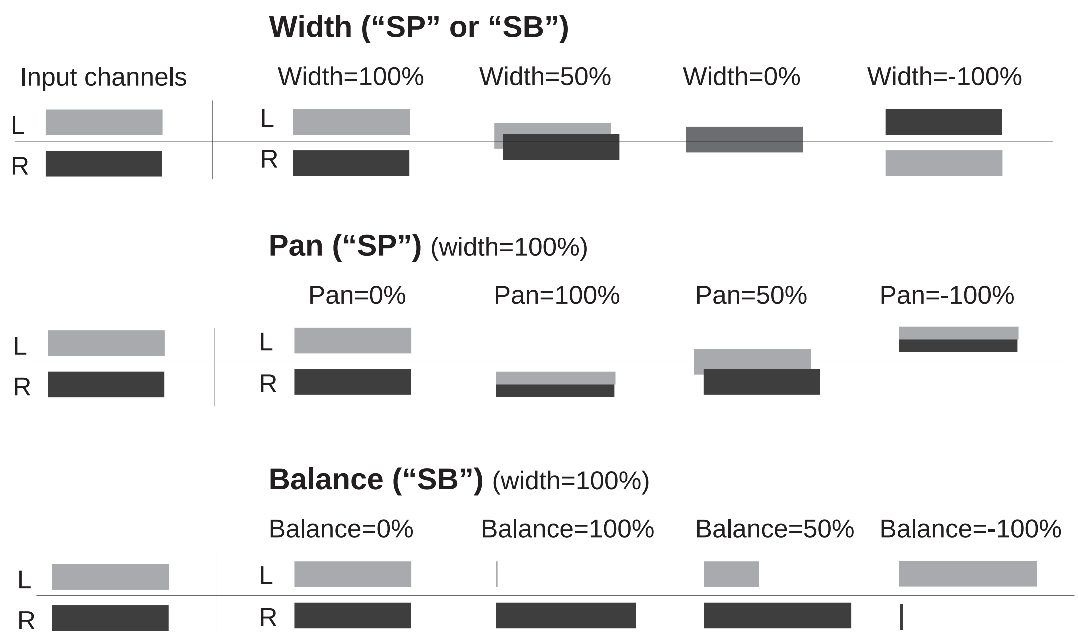


Figure 15-3 Width, Pan, and Balance controls

A Word About Gain

The Pan, Balance, and Width controls all have constant power curves, so that the combined signal level doesn’t change when you move the signals from side to side. However, if you use several Pans or Balances on a signal to keep the channels isolated throughout the entire signal chain (for example, if the Input send is panned 100%, and so are the Aux send, the Mix send, and the Mix output), you can increase the gain of the signal considerably.

Each stage of hard-panning adds 3 dB, so the increase in gain when the signal reaches the final output can be as high as 12 dB. In this case, you may want to trim the level at various stages to keep the signals from getting too hot.

Effects Buses

The four insert Effects buses (FXBuses) receive the signals from the Input Editor and process them. Press the **FXBUS** soft button to go to one of the FXBUS pages—the first time you do this after entering the Studio Editor, it will be the **FXBus1** page.To go to the other FXBUS pages, use the **Chan**/**Bank Up** and **Down** buttons. The number of the FXBus appears in the upper right corner.

The Arrow Meter

There are arrow meters on the FXBUS pages as well, right next to the number of the FXBus. These tell you when signal is coming into the bus, and also when signal is present *inside* the bus, so if you have a long reverb or repeating delay, for example, the arrow will keep flashing as long as the processing is going on.

FX Preset

The first parameter on an FXBUS page is the FXBus's FX preset. Set the cursor on it, and use the Alpha wheel to scroll through the FX presets currently in memory. If an FX preset name comes up in parentheses, for example, **(Really Big Plate)**, it means there is not enough processing power (PAUs) available at the moment to use this FX preset in this FXBus. We'll get to PAUs in a moment. **199 No Effect** is a "blank" FX preset, in which all signals pass straight through without any processing. It can be used as a starting point for creating your own FX preset. If you want to set up a "dummy" effects bus to pass signal directly to the Aux bus, use **0 None**.

Bus Outputs

The parameters on the right side of this page determine how the effected signal gets to the KDFX outputs. Each FXBus has four outputs, all of which are stereo:

- Its own dry (pre-effect) output
- Its own wet (post-effect) output
- The Mix bus
- The Aux bus

The output to the Mix effects bus has a level control with a range of **-79.0** to **+24.0 dB**, and an **Off** position. It also has a Balance control that works similarly to the Balance control on the inputs, by setting the relative levels of the two output channels. The signal is mixed with similar signals from the other FXBuses onto the Mix bus, which can be accessed on the OUTPUT page.

The output to the Aux bus has an identical pair of controls. Its signal goes to the global Aux effects bus, where it is mixed with similar signals from the other FXBuses, and then put through the Auxiliary Effects processor. From there it can be accessed on the OUTPUT page.

There is no external level control over the output of the FXBus itself—it just shows up, in pre-effect and post-effect versions, on the OUTPUT page.

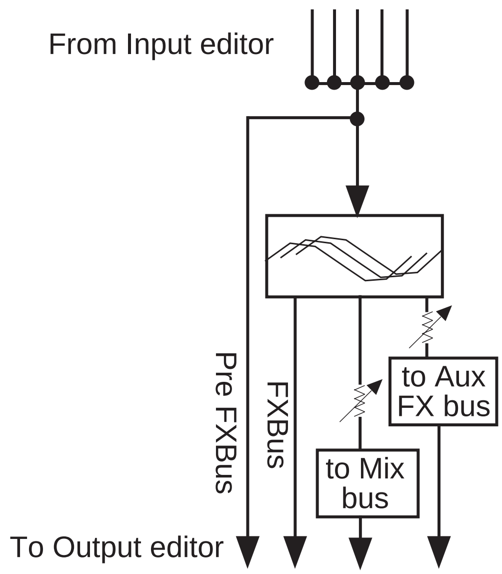


Figure 15-4 FXBus Signal Routing

The Aux Bus

The Auxiliary effects bus is accessed by pressing the **AUXFX** soft button on any page in the Studio Editor. It contains an FX preset, which is separate from those in the insert FXBuses—although it can be the same FX preset that is in use on one or more of the insert FXBuses. There is no Allocation parameter, because the Aux bus has a fixed allocation of 3 PAUs. Only a very few highly complex FX presets require more than 3 PAUs, so as you scroll through the FX presets here you won't see many names in parentheses.

Aux Bus Outputs

There are two outputs from the Aux bus: itself, and a feed (post-effect) to the Mix bus. The feed to the Mix bus, where it is combined with other Mix bus feeds from the four FXBuses, has level and balance controls. The Aux bus's own output has no post-effect controls, and goes right to the OUTPUT page.

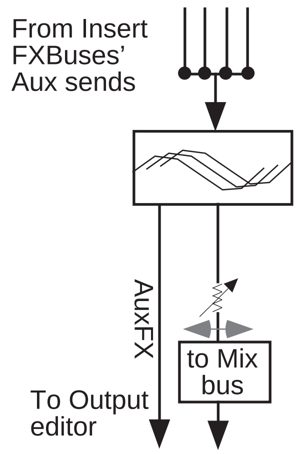


Figure 15-5 Aux Bus Outputs

About FX Presets

FX presets are where the processing takes place in KDFX. Each of the insert FXBuses and the Aux bus have their own FX preset, and they are all independent of each other. If you think of a studio as being the equivalent of a K2600 program, then an FX preset is the equivalent of a layer or keymap.

In any of the FXBuses, the FX preset is edited by placing the cursor on the name of the FX preset and pressing **Edit**.

The first parameter page, or PARAM1, appears. The algorithm that the FX preset is based on is selected at the top of this page. Highlight the name on the FXAlgorithm line, and scroll through the various algorithms. The algorithms are in the K2600's ROM (like ROM samples), and are not changeable by the user. As you change algorithms, the parameters displayed on the page change accordingly.

```
EditFXPreset:PARAM1          EffectSize:3/3
FXAlgorithm:1 Panaural Room
                    In Gain    :0.0dB
Wet/Dry    :30%     Out Gain   :2.0dB
Room Size  :15.2m
Pre Dly    :4ms     Decay Time:1.7s
HF Damping:14080Hz
<more  PARAM1 PARAM2                 more>
```

Notice also that as you scroll the algorithms, the EffectSize parameter in the upper-right corner changes. This parameter shows how many Processing Allocation Units (PAUs) the currently selected algorithm requires, followed by how many are available for this FX preset. If, for example, EffectSize is **2/3**, that means the algorithm requires 2 PAUs, and there are 3 PAUs available. More about PAUs soon.

Just below and to the right of the algorithm name is an Input Gain parameter, which adjusts the level of the signal coming into the FX preset from the input(s) sending to it. The relative level of the various inputs is determined on the Input pages, but you can change the overall level here—

for example, if you've combined several inputs into one FXBus and the signal is too hot for the FX preset, you can pad it down here. The Trim is adjustable from **Off**/**-79.0 dB** to **+24 dB**.

Below the Input Gain is usually (but not always) an Output Gain, which sets the level of the signal leaving the FX preset, which can be further modified by the Mix and Aux sends on the FXBUS page.

The other parameters that appear on this page are determined by the algorithm. Each algorithm has its own set of parameters, which may take up as many as four pages, accessed using the soft buttons **PARAM2**, **PARAM3**, etc. The parameters associated with each algorithm are discussed in detail beginning on page 15-35, and at the beginning of Chapter 10 of the *Musician's Reference*.

When you change a parameter on one of these pages, you have changed the FX preset, and if you want the change to be permanent, you must save the FX preset. Pressing either of the **<more>** soft buttons accesses Name, Save, Delete, and Dump pages for the FX preset. FX presets are stored in RAM, just like K2600 keymaps. When an FX preset is recalled, either by itself (from within the Studio Editor) or as part of a studio, its associated algorithm and all parameters are recalled with it.

There is another way to edit an FX preset's parameters without altering the FX preset itself, and that is by using bus overrides.

Bus Overrides (Bus Mods)

We haven't yet talked about the two parameters that are on the Insert FXBUS and AUXFX pages, directly underneath the name of the FX preset. These are called "bus overrides" or "bus mods," and they allow you to change parameters within an FX preset without actually going into the FX preset.

```
Wet/Dry     :35%
Out Gain    :2.0dB
```

For example, the bus overrides on FX1 are often Wet/Dry mix and Output gain. Normally, these parameters would be found inside the FX preset, and if you changed them, you'd have to save the new FX preset in order to keep the changes.

Instead, using bus overrides, you can adjust these two parameters and hear what they sound like while you are adjusting them *without* going into the FX preset. When you save the studio, these parameter values are saved, but they are not part of the FX preset—they are part of the *studio*. Therefore, the FX preset remains unchanged (and if the FX preset is in use elsewhere, it hasn't changed there), but these two parameters *in this particular FXBus* have been altered.

Wet/Dry and Out Gain are the default bus overrides you will encounter most often, but in some algorithms and FX presets, other parameters are accessed as bus overrides. For example, on some compressor algorithms, the first bus override is an In/Out switch; and on some dual-channel delay and filter algorithms, the overrides are separate Wet/Dry controls for the left and right channels.

Making and Breaking Bus Overrides

Some studios supplied with KDFX, when you first encounter them, have the bus mods in place, but they are not engaged—that is, they're not actually overriding anything, but instead simply show the values of the corresponding parameters inside the FX preset unchanged. To see this, choose an FX preset on the FXBUS page and look at the values of the override parameters, **Wet/Dry** and **Out Gain**. Now go inside the FX preset by highlighting the FX preset's name and pressing **Edit**, and find those two parameters on the first parameter page (PARAM1). You'll see their values are the same as on the FXBUS page.

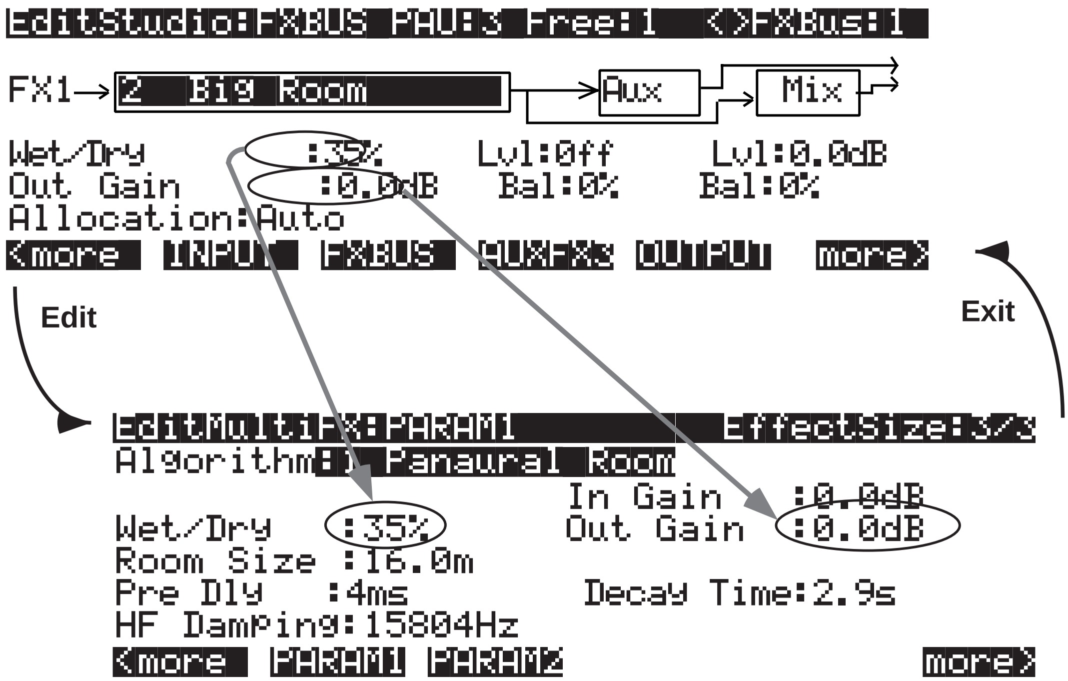


Figure 15-6 Bus Overrides in Place, But Not Engaged

Press **Exit** to go back to the FXBUS page.

To engage a bus override, you have to use it, which you do by moving the parameter away from its nominal value on the FXBUS page. Do that, and then go back inside the FX preset.You'll see that the value of the parameter you've changed is now shown as **BusMod**, meaning that the FX preset parameter is under the control of the bus override.

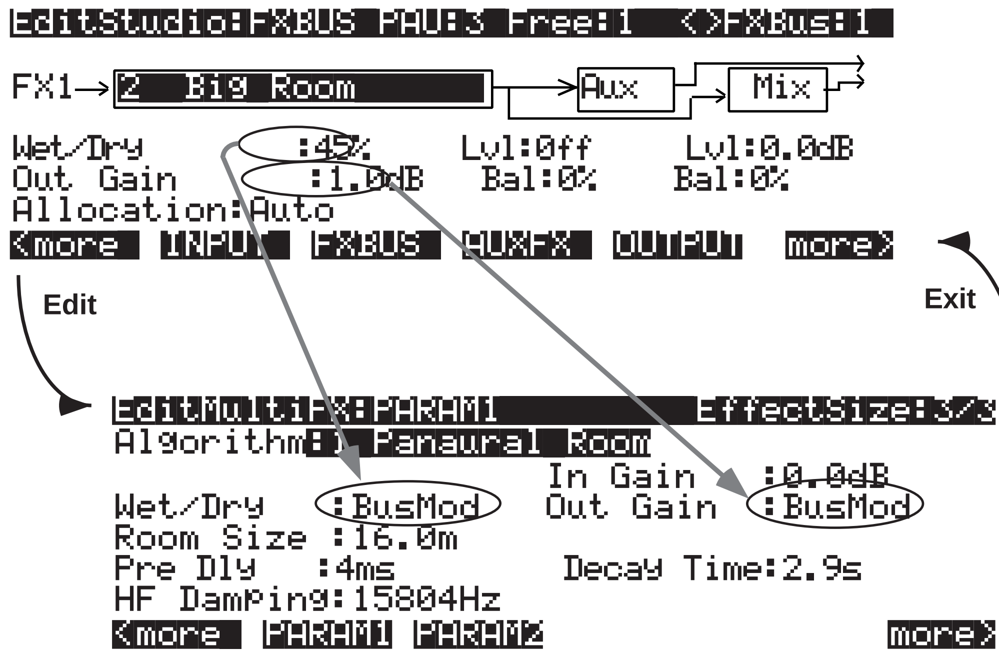


Figure 15-7 Bus Overrides Engaged

If you now change the parameter value from *inside* the FX preset, it *breaks* the bus override. By doing this, however, you have changed the FX preset, and when you leave the FX preset you will be asked if you wish to save it. If you then save the FX preset, the bus mod will be permanently broken, but if you *don't* save the FX preset, it remains in its previous state, which means the bus mod *remains intact*.

Since bus overrides are part of the studio, they are not saved permanently until the studio is saved.

Changing the Bus Overrides

The default bus overrides that come up when you are working in a studio are often convenient, but you aren't required to use them as they are. You can assign *any* two of the FX preset's parameters to the bus overrides. (You cannot, however, assign both bus overrides to the same parameter.) When you save the studio, the parameters you've selected for the bus overrides on each FXBus are saved, along with their values.

If you set up a bus override and adjust a parameter's value, and then change your mind and assign a *different* parameter to that override, the value of the first parameter (the one no longer assigned to a bus override) will revert to its original value—that is, the value set *inside* the FX preset. This can be a helpful feature, in that it means you can use bus overrides as a "window" into an FX preset by scrolling through the various parameters. But keep in mind that you can't use this method to tweak parameters, because as soon as you move on, the parameter you tweaked gets "untweaked."

Allocation

Understanding PAU allocation and how it works is very important for using KDFX to its fullest potential.

What's a PAU?

The basic unit of signal processing in KDFX is the Processing Allocation Unit, or PAU. There are a total of 4 PAUs that can be split among the four insert FXBuses, and another 3 PAUs for the Aux bus. These two sets of PAUs are distinct and are not interchangeable.

The number of PAUs on an FXBus is determined by the algorithm inside its FX preset. Different algorithms require different numbers of PAUs, as shown in the Size parameter at the top of the screen when you are on the FXBus edit page:

```
EditStudio:FXBUS Size:2
```

or inside an FX preset:

```
EditFXPreset:PARAM1    EffectSize:2/1
```

A simple delay, flanger, compressor, exciter, or small reverb uses only 1 PAU. A complex phaser, pitcher, or multiband tone control uses 2 PAUs. A really complicated reverb or graphic equalizer may use 3 PAUs. Only a handful of really wild algorithms use 4 PAUs.

The Allocation parameter on each FXBUS page determines how many PAUs are available for the FX preset assigned to that bus. If the parameter is set to **1**, then only FX presets that use algorithms requiring 1 PAU will be available for the bus. If you try to assign an FX preset that requires more PAUs, its name will show up in parentheses, and the sound will pass through the bus unprocessed.

If you know you want a certain FX preset in a particular FXBus, you can select it, and then set the Allocation parameter for that bus to match the PAU requirements of the FX preset.

A PAU is a Terrible Thing to Waste

Be careful not to set the Allocation parameter *too* high. If you set it on some bus to 3, for example, and you are using an FX preset on that bus that requires only 1 PAU, the other 2 PAUs are being wasted, since they are not available to be assigned to other buses. The other buses' Allocation parameters will not go higher than 1, and therefore no FX presets whose Size is greater than 1 can be selected for any of them.

PAUs are normally allocated on a first-come, first-served basis. If you set one FXBus to a PAU of **3**, then you will be able to set the Allocation on the next bus you set to **0** or **1** only. If you then set that second bus to **1**, then you won't be able to set a value greater than **0** for *any* of the other buses.

If you know that you won't be using an FXBus, or that you'll be using it only as a "dummy" to route signals somewhere else, you can set its Allocation to **0**.

Auto Allocation

On any of the buses, you can set Allocation to **Auto**. **Auto** means that the PAU allocation for that bus will automatically adjust itself to the currently selected FX preset. However, **Auto** obviously cannot create PAUs when they are in use elsewhere, and Allocations that have been manually set take precedence over Auto Allocations. So for example, if you set the Allocation of FXBus1 to **3**, and set the Allocation of FXBus2 to **Auto**, the maximum number of PAUs available to FXBus 2 is still only 1, and if you try to load a Size-2 FX preset into FXBus2, it won't work and the FX preset's name will show up in parentheses.

If *all* buses' Allocations are set to **Auto**, then PAUs are not allocated first-come, first-served, but instead are allocated in *numerical* bus order: if an FX preset requiring 3 PAUs is loaded into FXBus1, then only 1 PAU will be available for the other buses, regardless of which FX preset got assigned to which bus first.

A parameter called Free appears at the top of every FXBUS page, telling you how many PAUs in the current studio are unallocated and available.

Effect Size

While you're editing an FX preset and selecting algorithms for it, the EffectSize parameter at the upper right of the EditFXPreset page informs you of the PAU situation. The first digit in this parameter's value is the number of PAUs the currently selected algorithm requires, and the second digit is the number that have been allocated to the bus, either manually or automatically. If the first digit is larger than the second, the algorithm is not available, and if you choose it, its name will show up in parentheses and the sound will pass through unprocessed—just like an unavailable FX preset on an FXBUS page.

If the FXBus's Allocation is **Auto**, when you change the algorithm inside the FXPreset Editor, *both* digits of the EffectSize display will change. If you call up an algorithm that requires more PAUs than are currently available, the second digit will change to **0**—since the algorithm can't be loaded, the PAUs are freed up for use elsewhere.

Designing with PAUs

One simple way to use Allocations when designing a KDFX studio is to put all of the buses in Auto, and start with your most complex processing on FXBus 1, then assign other FX presets to the other buses as they are available.

PAUs on the Aux Bus

The Aux bus is a whole separate processor with 3 of its own PAUs. Allocation is not an issue with the Aux bus, since its 3 PAUs are assigned to it permanently. (There is a fourth PAU in the Aux bus, but it is used for mixing and routing, and so it's not available.)

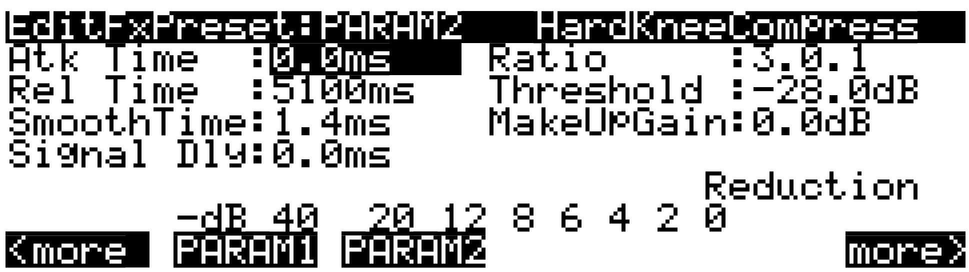
Any FX preset with a PAU requirement of 1, 2, or 3 can be used. When editing the Aux bus, there's no need for an Allocation parameter or a Free parameter. However, when you are scrolling through algorithms, the EffectSize parameter will be displayed, with its second digit always **3**. Should you try to access an algorithm on the Aux bus that requires 4 PAUs, the name of that algorithm will be displayed in parentheses.

The Aux bus is generally used as a global processor, but it doesn't have to be. You can set up one FXBus as a "dummy"—set its FX preset parameter to **0 None**—and send its output to the Aux bus, and at the same time turn off the Aux sends from all the *other* buses. Now the Aux bus can function as an insert FXBus, with 3 PAUs all its own. So if you run into trouble with PAUs on the insert FXBuses and can sacrifice a global effect, this is one solution.

Metering

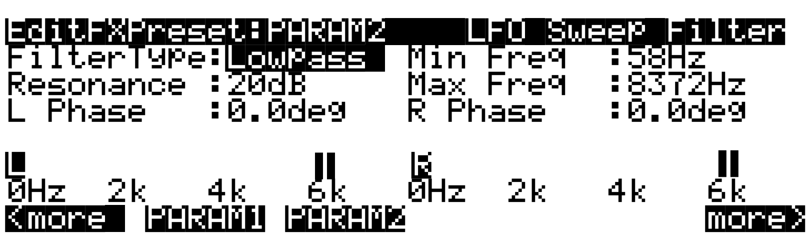
All Input and FXBUS pages have rudimentary signal meters (the arrows), as we've seen. On the Input pages, the arrow next to the input letter flashes when there is signal present at the input. On the FXBUS and AUXFX pages, the arrow next to the FXBus number flashes as long as there is signal being processed—in effect, it's an output meter. The arrows flash whenever the signal level exceeds 14 bits below full scale, which is -84 dB relative to the maximum level the KDFX can handle. Since typically the K2600 operates with about 20 dB of headroom, this translates to about -64 dB relative to normal operating level

In addition, a number of KDFX's algorithms include a fast visual metering system for monitoring what the algorithms are doing. Many of these algorithms deal with dynamics control like compression and limiting. An example is Algorithm **950 HardKneeCompress**. Meters are most often found on the PARAM2 page of the FXPreset Editor:



In this algorithm, the meter is showing the gain reduction that the compressor imposes on the signal coming into it.

Metering is also used for other purposes in some other algorithms, for example **902 LFO Sweep Filter**. In this algorithm, the meters show the current center frequency of the left and right filters as they sweep up and down the spectrum:



Meters use up a certain amount of KDFX's processing power, although less than a whole PAU, and so they are only used in algorithms where extra power is available. If an algorithm would need to increase its Size—say, from 2 PAUs to 3—with the addition of metering, then metering has not been included in that algorithm. This is to ensure that the maximum amount of processing power is available for actual signal processing.

Output Section

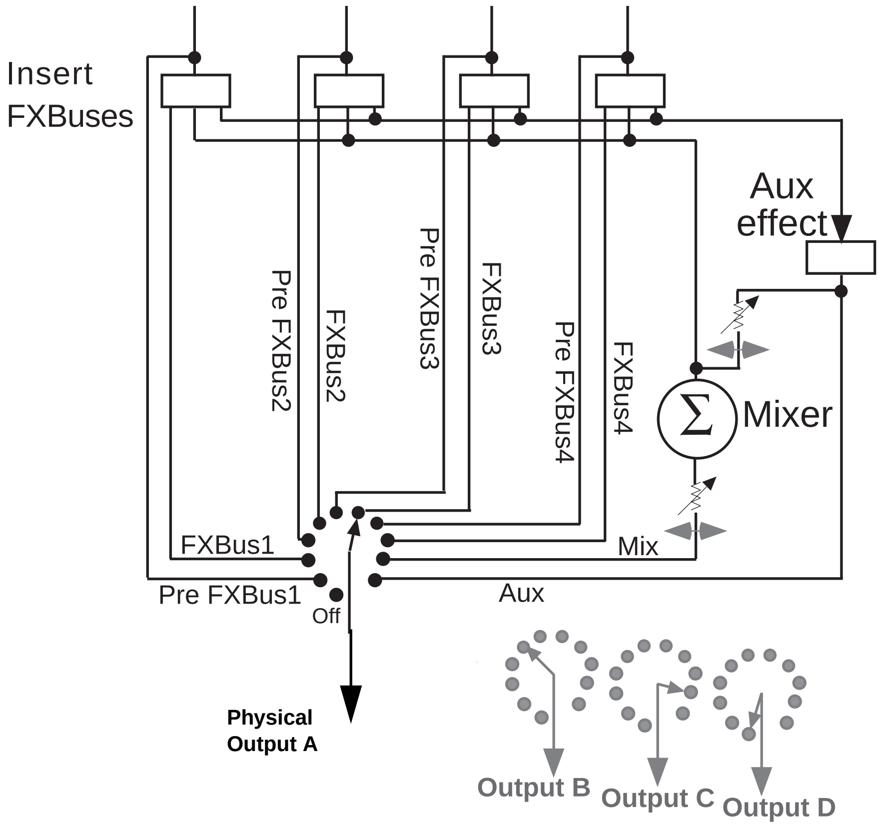


Figure 15-8 FXBuses and Physical Outputs

In the Output section, the various signal paths are routed to the K2600's physical outputs. Each of the four output pairs, A, B, C, and D, has a selector switch to determine which signal it will carry. All signals are stereo. The choices are:

Off No output.

PreFX*n* The signal coming *into* the FX preset on FXBus1, 2, 3, or 4. All of the EQs and pan/width/balance settings of the input modules that are assigned to FXBus*n* are active on this signal path, but the FX preset is not.

FXBus*n* The direct output, post-FX preset, from any of the four insert FXBuses.

Mix The signal from the Mix output, which can include the sum of any or all of the insert FXBuses, and/or the Aux bus. The Level and Balance parameters on this page control this output.

AuxFX The outputs of the Aux bus.

There are no restrictions on the settings. If you like, all four outputs can be carrying the same signal: PreFX1, for example.

Separate Analog and KDS Outputs

The K2600's separate analog outputs (four pairs, A–D) are wired in parallel and are "live" at all times. So are the eight digital outputs (also four stereo pairs) available with the digital I/O (KDS) upgrade option. The parameters on the OUTPUT page affect both the separate analog and KDS outputs.

Analog Mix Output

The K2600's analog Mix output combines the four output pairs into a single stereo analog pair. Each of the analog outputs, A–D, carries the signal from the KDFX bus specified for each output pair on the OUTPUT page of the Studio Editor. The signal at the Mix output always carries the summed signal of outputs A–D; connecting cables to Outputs A–D does not remove the corresponding portion of the signal from the Mix outputs.

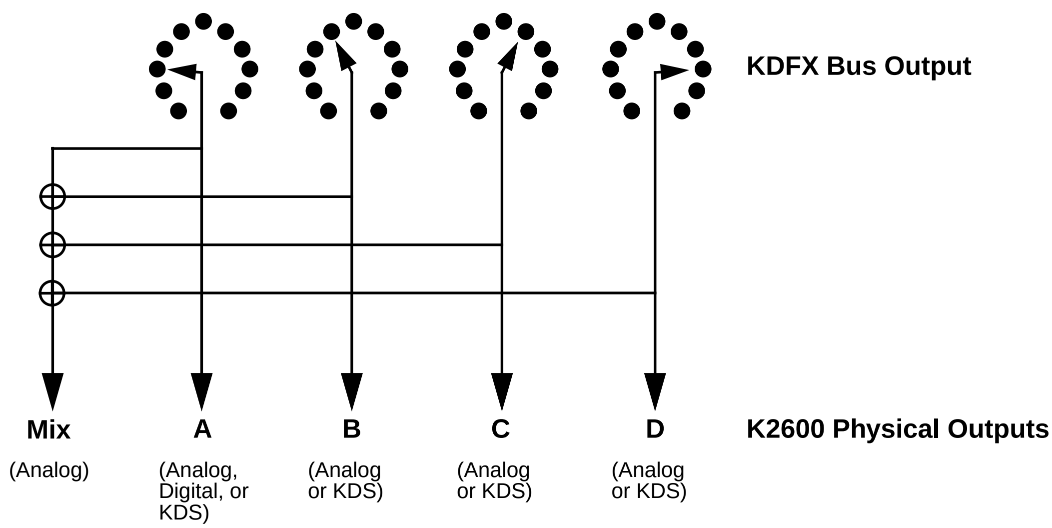


Figure 15-9 K2600 Analog, Digital, and KDS Outputs

Digital Output

For digital output, you'll need either the sampling option or the digital I/O (KDS) option. In either case, the digital output of the K2600 (a single stereo pair) is wired in parallel with output pair A. It gets whatever signal is routed to Output A on the OUTPUT page of the Studio Editor.

With the digital I/O option, you can use the XLR or optical outputs, or the separate KDS outputs (although this requires the Kurzweil DMTi digital interface for full functionality). With the sampling option, you can use the XLR or optical outputs.

The sampling option also enables you to sample the K2600's own output.

1. On the Sample-mode page, set the Input parameter to **Analog**, and the Src parameter to **Int**. This routes the signal at the K2600's Mix outputs to the analog sampling input, ignoring the external sampling inputs on the rear panel.

2. Set the Input parameter to **Digital**, then set the Out parameter, if available, to **A/D**. This causes the digital outputs to mirror the signal coming from the K2600's Mix outputs. (Set the Out parameter to **Dir** to mirror the Output A signal at the digital outs.)

***Note**: If you don't have the Digital I/O option, you won't see the Out parameter, which means that the digital output mirrors the K2600's Mix outputs.*

When the Input parameter is set to **Analog** and the Src parameter is set to **Ext**, the signal that appears at the digital output is the signal coming into the K2600's *external* analog inputs, and the K2600 is acting as a very fancy, very high-quality A/D converter.

Value of Out Parameter	Value of Src Parameter	Signal at Digital Output
Dir	N / A	KDFX Output A
A / D	Int	K2600 Mix Output
A / D	Ext	K2600 Analog Input

Table 15-1 Digital Output Switch Settings

Saving Studios and Other objects

Saving

Saving a studio or an FX preset is handled the same as any other K2600 object—see *Saving and Naming* on page 5-3 if you need help with this.

To save an object, from any of the editing pages within the object, press **<more** or **more>**, until you see the file-management pages:

<more Name Save Delete Dump more>

Name lets you rename the object, using the standard K2600 naming window. **Save** lets you save the object to its present slot or to a different one, and also gives you the option of renaming it before saving it, or accessing the Object utilities—see Chapter 13.

Delete deletes the object from RAM. If it is in a slot where there is a ROM studio or FX preset, then the slot will revert to the ROM studio or FX preset.

Dump performs a MIDI System Exclusive dump of the object over the MIDI Out cable for external storage.

The Save option is automatically invoked if you have done any editing within an object and you press **Exit**.

Do I Always Have to Make a New Studio?

While it may sound as if every time you make a change in a studio parameter, you have to create a new studio to save it, this is not necessarily true. Studios do take up room in program RAM, and although the number of studios that can be maintained inside the K2600 is large, it is not unlimited. Besides, it can be very cumbersome keeping track of a large number of studios which have only small differences between them.

A way to avoid this problem is to use Program or Setup mode, and use FXMods—studio-parameter controls that live inside a program or setup—to make changes in the studio parameters. When you load the program or setup, the studio changes accordingly. This means you don't have to save each variation in the studio as a separate studio; the variations live in the program or setup.

See *Real-time Control of KDFX* below for more information.

Disk Functions

Studios and FX presets are loaded from and saved to disk exactly the same as other K2600 objects. FX presets are dependent objects within studios, so when you save a studio, you are given the opportunity to save the dependent FX presets as well. (Algorithms are in ROM, so they don't need to be saved.) Again, see Chapter 13 if you need help.

Real-time Control of KDFX

Studios can be static, but they don't have to be. One of the most powerful features of KDFX is the ability to change any of the operating parameters in a studio in real time. Mix levels and pans, EQ values, effects parameters, and almost any other settings can be controlled dynamically and smoothly, giving KDFX a high degree of flexibility in performance and automation.

KDFX parameters can be controlled from the following

- The K2600's sliders, wheels, ribbons, and pedals
- External MIDI sources like another controller or a sequencer
- Internal functions like LFOs, envelopes, and FUN generators

Real-time control of a studio is called effects modulation, and a link between a program or setup and a studio parameter is called an FXMod.

Linking to Programs and Setups

If you're familiar with real-time control of the K2500's original internal effects, you'll be happy to know that real-time control of studios works the same way.

Real-time KDFX studio control normally originates within a program or setup. For the link between a studio and a program to work, the FX Mode parameter on the Effects-mode page must be set to **Program**. If you want to link a studio and a setup, the FX Mode parameter on the Effects-mode page must be set to **Setup**. Setting the FX Mode parameter on the Effects-mode page to **Auto** means that it will follow whatever mode the K2600 is in—Program or Setup—which can be particularly useful when you are switching between programs and setups in a Quick Access bank, or when you just don't want to worry about which mode you're in.

```
EffectMode      Xpose:0ST       <>Channel:1
FX Mode:Program
FX Chan:1


Dither :Medium       DigOut :16 Bit
Octav- Octav+ Panic         Chan-  Chan+
```

Notice that when FX Mode is set to **Program** or **Setup**, the studio is not shown, as it is when the FX Mode is set to **Master**. So you can't go into a studio and edit it from this page—you have to go through the program or setup, from its KDFX page.

Note: When FX Mode is set to **Auto**, then FX Mode follows the K2600's operating mode—it's either **Program** or **Setup**. When you are on *this* page, however, FX Mode is actually **Master**. But don't screw up your brain thinking about this—we'll get back to it later in this chapter (*KDFX in Auto Mode* on page 15-34).

For the sake of clarity, during most of this chapter we will talk about linking KDFX only with programs. The procedures for linking KDFX to a setup are essentially identical, except for the setting of the FX Mode.

The KDFX Pages

If you look inside any program, you will see several pages that handle the studio and KDFX control assignments. Open up a program with the **Edit** button and press **more>** until you see this at the bottom of the display:

```
<more  KDFX  FXMOD2 FXMOD3 FXMOD4 more>
```

These are the FXMod setup pages, and in fact there are eight of them—press **more>** again to see the others.

```
<more  FXLFO FXASR FXFUN ImpFX  more>
```

Go back to the first set (press **<more**), and press the **KDFX** soft button to look at the first of the FXMod setup pages:

```
Edit.Prog:KDFX                     All Layers
Studio:278 Hall+flange+rm

Bus: Param:      Adjust:   Source:   Depth:
FX1  Wet/Dry     12%wet    MWheel    40%wet
FX2  Fdbk Level  25%       MIDI23    60%
InA  EQ1 Bass G  0.0dB     MIDI24    21dB
<more  KDFX  FXMOD2 FXMOD3 FXMOD4 more>
```

The top line tells us we're in the KDFX section of the Program Editor. The second line shows the studio that is linked to this program. Any changes made on this page do not directly affect the studio, they affect only this program's *control* over the studio.

The last three lines (ignoring the Soft buttons) show us which FXMods are active in this program: one is controlling Wet/Dry mix on insert FXBus1, one is controlling feedback level on insert FXBus2, and one is controlling the Bass Gain on the first EQ of Input A.

Looking Into the Studio

You can go into this studio, to see what the parameters are doing—a good idea when you are setting up FXMods so you understand them in context—or to alter a fixed parameter. You can do this without leaving the Program Editor: highlight any parameter on this page and press **Edit**. When you are done with the studio, pressing **Exit** will bring you back to this page. If you have made any changes in the studio, you will be prompted to save the studio, and if you don't do so, the changes will be disregarded.

If you do save the studio, either in the same numbered location or a new one, the new studio will now be linked to this program. If you have changed the studio's number, then you'll also have to save the program when you leave the Program Editor, so that the program knows which studio to link to the next time you call it up.

Setting Up FXMods

The KDFX page allows three different FXMod control assignments to be made. More FXMod assignments are available on the FXMOD2, FXMOD3, and FXMOD4 pages, each of which has five setups. This gives a grand total of 18 studio parameters that can be under real-time control. Don't worry, you don't have to use them all.

Bus Assignments

The first column lets you choose which bus inside the studio you want this FXMod to control: Input A, B, C, or D (if any of the Inputs is set to Mono, you will get to choose individual channels, for example A/L and A/R); FXBus 1, 2, 3, or 4; the Aux bus; or the Mix bus.

Parameter Assignments

The second column chooses the specific parameter on the selected bus. Scrolling through the choices shows that this selection is context-sensitive: it shows only parameters that are being used in the current studio on the selected bus, so you can't make assignments to irrelevant or nonexistent parameters. It "knows" which EQs are active, and what modes they are set to; which Input sends are assigned and whether the sends are in Pan or Balance mode; what parameters are being used in the FX preset on the particular bus; etc. If you need to know more about how the studio is set up, you can go into it and look around, as explained above.

Here's an example: set the FX Studio to **201*RngMd/PFD/Plt** which we looked at on page 9-16. Set the bus on the first line to **InA**. Set the cursor under Param: and scroll the choices. They correspond exactly to the parameters available in the Input Editor page: level and frequency for the two EQs, and SendLvl, Pan, and Width for the two FX sends. Note that you *cannot* change the FXBus assignments on an input from here; you must do that within the studio itself.

Change the bus to **FX1**, and all of the parameters from the FX preset on FXBus1 are available for selection, including Mod Mode and the various settings for the preset's internal oscillators, as well as the bus's output controls: Mix Level, Mix Balance, Aux Level, and Aux Balance. Change the bus to **AuxFX**, and the Aux bus's parameters are available: levels, delays, room types, etc.

Setting the bus to **Mix** makes available the Level and Balance controls from that page.

What Can't Be Controlled

You cannot change any parameters through KDFX that would involve a major reconfiguration of the studio:

- Bus assignments on the Input pages
- Selecting FX presets on the FXBus pages
- Allocation on the FXBus pages
- Selecting algorithms within the FX presets
- Bus assignments on the Output page

If you want to be able to change any of these in real time, you will have to create a new studio, link it to a different program or setup, and then call it up using a Program Change command.

In addition, there are a few parameters that can cause serious glitching if they are changed in real time. The most common of these are the "Room Type" settings in reverb algorithms. While there is nothing in the software to prevent you from assigning an FXMod to Room Type, you need to be aware of the potential consequences. See *Static FXMods* on page 15-27.

Adjust

The Adjust setting is the starting value of the selected parameter when it is under KDFX control, similar to the entry value of a controller in a setup. This might very well be different from the value of the parameter when the studio is *not* under FXMod control, so don't get confused. If you are in Program mode, and this is the current program, the Adjust value takes precedence over the studio's fixed value, and it's the Adjust value that will be called up when you call up the program.

Source

The Source parameter determines which real-time control—internal, MIDI, etc.—is going to affect the selected studio parameter. As with all K2600 real-time controls, the range of control sources is very large:

- OFF (the parameter is not affected by any source and stays at its Adjust value)
- ON (the parameter is set to the maximum value determined by adding the Adjust and Depth values)
- MIDI Continuous Controllers 1-95 (*see Note*)
- Channel State
- Pressure
- Pitch Wheel
- The usual list of controllers, as described in Chapter 4 of the *Musician's Reference*: ASRs, FUNctions, Clocks, LFOs, Internal Controls, Random Generators, etc.

***Note**: Under some circumstances, particularly when the K2600 is in Setup mode, there are certain restrictions on which MIDI sources you can use.*

Dedicated FXMod Control Sources

There are a few control sources that apply exclusively to FXMods:

- FXLFO1, FXLFO1ph, FXLFO2, and FXLFO2ph—two LFOs and their phases.
- FXASR1 and FXASR2— two three-stage (Attack/Sustain/Release) envelopes with selectable triggers and Normal, Hold, and Repeat modes.
- FXFUN1, FXFUN2, FXFUN3, and FXFUN4—Functions. Yes, more Fun with KDFX!

The ASRs, FUNs, and LFOs work exactly the same way they do in any other part of a program, except these are extra control sources for use *only* with FXMods, and are not available for other program functions. They are global for all of KDFX—you can't apply these controls to just one FXBus.

The parameter values for these controls are saved with the program. As we saw earlier, you get to the pages for the FXMod control sources by pressing the **more>** soft button until you see these soft buttons:

<more | FXLFO | FXASR | FXFUN | ImPFX | more>

In the Program Editor, you can also get to these pages directly from the KDFX page or one of the FXMOD pages: select one of those parameters as a source, and then press **Edit**.

Tempo-based Parameters

There are several different ways KDFX can respond to tempo information, from the internal sequencer or an external one. These are discussed later in this chapter (see *Tempo-based Control of KDFX* on page 15-32).

Depth

The Depth parameter lets you specify a range of change in values that the real-time control will make, using the Adjust value as a minimum or starting point. This range can be positive or negative, and the values are displayed in the context of the studio parameter: seconds, dB, %, Hz, cents, etc.

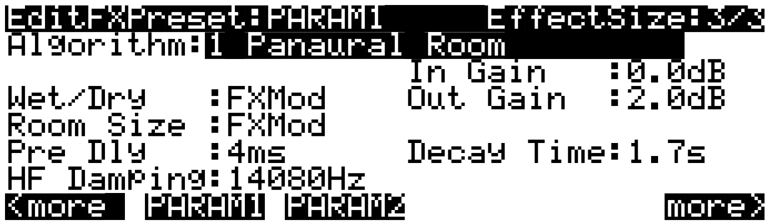
At the maximum setting of the Source (for example, Mod Wheel all the way up), the value of the parameter = Adjust + Depth. So if the parameter is Out Gain, the Adjust is 1.0 dB, the Source is Mod Wheel, and the Depth is 4.0 dB, then at the Mod Wheel's highest point, the output gain will be 5.0 dB.

For Source values less than maximum, the formula is: parameter = Adjust + (Depth x Source), where the Source is considered to be varying between 0 and 1 (or in some cases, such as Pitch Wheel, between -1 and +1). So using the same example, when the Mod Wheel is halfway up (MIDI value 64), the gain is 1.0 + (4.0 x 1/2) = 3.0 dB, and when it is all the way down (MIDI value 0), the gain is 1.0 + (4.0 x 0) = 1.0 dB.

The formula works the same way for negative Depth values. Given the same example, but with a Depth of -4.0dB, at the Mod Wheel's minimum point, the gain will be 1.0 dB; at its halfway point it will be -1.0 dB (1.0 + (-4.0 x 1/2)); and at its maximum point it will be -3.0dB.

Showing Who's in Control

When you are in Program or Setup mode and you look inside the current studio or its FX presets, any parameters that are under FXMod control will not display numerical values, but instead will say **FXMod**. Don’t touch any of those parameters for now—we’ll explain why in a moment.



Note that if you look at the studio when FX Mode is in Master, the FXMods will not be displayed, because the FXMods are not in effect! So don’t get confused.

Breaking the Links

When FX Mode is **Program** (or **Setup**) and you are in the Studio Editor, if you change a parameter that has been assigned an FXMod, it breaks the link—the parameter is no longer under FXMod control, but is now a static value. If you now save the studio (either in the same location or a different one), the link stays broken. However, if you leave the studio *without* saving it, and go back to the FXMod pages of the Program Editor, the link automatically reestablishes itself.

If you change the FX preset on an FXBus, the FXMod links to the preset on that bus may or may not break. If the preset you’re calling up uses *the same algorithm* as the preset you’re leaving, the links will stay intact. If it uses a different algorithm—even a similar one—the links will break. On the other hand, if you are inside a preset, and you change its algorithm, that breaks the links. Again, the links will be reestablished if you leave the preset without saving it.

Links to the Aux and Mix parameters on an FXBus do not get interrupted when you change the FXpreset on that bus.

The same rule applies when you are on the KDFX page in a program or setup and you change the studio. Links to the Input and Output pages and the Level parameters on the FXBus pages will be retained, but links to processing parameters will be retained on a given FXBus only if the algorithm inside the preset on that bus isn’t changed.

What About Bus Overrides?

Bus overrides are transparent, as far as KDFX is concerned. Any FXMods that involve a parameter inside an FX preset are set up directly between the program and the FX preset itself. If the parameter happens also to be assigned a bus override, it doesn’t make a difference to KDFX—the program will control the parameter as if the bus override wasn’t there.

However, bus overrides are still active *within a studio*, and changing a bus override value in a studio will, as usual, change its associated FX preset parameter. If there happens to be an FXMod *also* controlling that parameter, *the FXMod link will be broken*, just as if you reset the parameter from inside the FX preset.

Using FXMods So You Don’t Have to Change Studios

We’ve mentioned that FXMods are a good way to get around the need to create a new studio every time you want to make a small change in a studio’s parameters. The same studio can be used for many different purposes if you use FXMods to control it. FXMods can set the gain of the signals going to the various FXBuses, set panning and output levels of the signals from the buses, configure the mix going to the Aux bus, and even turn FXBuses on or off, as well as set processing parameters.

Since any parameter except the ones that reconfigure a studio (see *What Can't Be Controlled* on page 15-24) can be under FXMod control, the amount of variation between the various program- or setup-controlled versions of a single studio can be very great.

Static FXMods

Static FXMods, that is, FXMods that are just going to be used to "set and forget" studio parameters when a program or studio is called up (as opposed to dynamically controlling them), are easy to set up. On the FXMOD page, select the bus and parameter you want to control, set the Adjust value to the parameter value you want, and set Source to **OFF**. When you are done, save the program or setup normally. Now whenever you select the program or setup, the parameters in the FXMods will be immediately reset to the values you've specified.

Room Type and other potentially glitch-producing parameters work much better as static FXMods than they do under dynamic control. If you need to change a Room Type in a reverb when you change a program, you can do so with a static FXMod without hearing horrible glitches, as long as you make sure there is no signal passing through the reverb at the moment of the program change.

Importing Studios From Another Program or Setup

If you have a killer studio with FXMods set up in one program (or setup) and you want to use the same studio and FXMods in a different program, you don't have to rebuild the studio by hand: you can copy studios between programs or setups in one operation.

Press the **ImpFX** soft button, and you'll see this:

```
Edit Program:Import KDFX

From Program  200*My Orchestral Studio
     (Studio 41 auxChorTube Plt   )
Prog  Setup                  Import Cancel
```

Use the soft buttons at the left to select either a program or a setup as the source for the studio and FXMods you are importing. Select the number of the program or setup you want to import from. As you change the program or setup, the studio associated with each program or setup will display in parentheses below the program or setup.

The studio will also kick in, so you can hear it affecting the program you're listening to.

When you've chosen a program or setup to import a studio from, press the **Import** soft button. "KDFX from…imported" will flash on the screen, and you will be returned to the Program Editor of the program or setup you are working on.

Studios from setups can be imported into programs without restriction, and vice versa.

FXMods on Imported Studios

If you have imported a studio into a program, the FXMods that were in the program you've imported *into* are now all gone. They are replaced with the FXMods associated with the program you imported *from*. It doesn't matter if there were more FXMods in the old program than in the new—even if the new program has *zero* FXMods, the old ones are all erased.

Using KDFX Live From the K2600 Keyboard

The sliders and wheels on the K2600 keyboard, and the pedals connected to the keyboard, can be extremely useful with KDFX, if you assign FXMods to the K2600's controllers. The ribbon, for example, can be used in a "Pitcher" FX preset to change the pitch of the signal, while sliders and pedals can be used to control reverb time, flanger feedback, EQ, or any of the myriad parameters available in KDFX.

KDFX in Setup Mode

To use KDFX with a setup, make sure that FX Mode (on the Effect-mode page) is set to **Setup** or **Auto**.

Like programs, every setup has a set of FXMod routings associated with it. These are identical to the routings available in programs: there are four FXMod pages, three dedicated FX function pages, and an Import FX page. The procedure for setting up the routings is the same as in a program. KDFX settings can be imported into a setup from either another setup or a program.

When KDFX is under setup control, any FXMods in the programs *within* the setup are ignored. If you have a program that contains a studio and FXMods that you would like to use while playing a setup, import the KDFX from the program into the current setup.

MIDI Control and Setup Mode

You'll notice that when you set FX Mode to **Setup**, FX Chan goes to **None**, and can't be changed. The MIDI receive channel for controlling KDFX is not determined here, it's determined in the setup itself: it's the channel used by Zone 1 of the setup. Any incoming MIDI commands on other channels, while they may play *sounds* in the setup, will not affect KDFX parameters.

If for some reason you don't want KDFX to be in Setup mode while you are playing a setup—for example, if you want it to be under control of an external MIDI device and not change studios when you change setups—you can set the FX Mode to **Program**. In this case the *program that is in the zone assigned to the FX Channel*, (which may or may not be part of the setup) will control KDFX.

So for example, say a setup has three zones, which are assigned to channels 2, 4, and 6. If FX Chan is set to **6**, then the studio (and FXMods) assigned to the program in Zone 3 will be active. If none of the zones' channel assignments match the FX Channel—say it's set to **16**—then nothing you do on the K2600 will control the studio and FXMods. However, an external MIDI source, like a sequencer, sending on channel 16 *will* control the studio and FXMods, and the program that is on channel 16 will determine what they are.

This technique can be useful when you want to have a sequencer control KDFX, while you are also playing along on the K2600 keyboard in Setup mode.

KDFX in Program Mode

If you are playing the K2600 in Program mode, then you must make sure that the current channel of the K2600 (at the upper right of the Program-mode page) agrees with the FX Channel (on the Effects-mode page). Otherwise the local keyboard commands, while they will control the other current program parameters, will not address KDFX. If FX Chan is set to **Current** then you don’t have to worry about this.

Performance Modes and Effects Control

Here’s how to tell what’s controlling what, depending on the various modes. See *FX Channel* below for information about the MIDI channel for external control of KDFX. Note that you can use SysEx commands to control KDFX regardless of the setting for FX Mode.

K2600 Mode	FX Mode Value	What Sets Physical Control Assignments (Sliders, Wheels, etc.)	What Controls KDFX	MIDI Channel for Control of KDFX
Program	Program or Auto	Control setup	Program’s FXMods	FX Chan
Setup	Setup or Auto	Setup	Setup’s FXMods	Channel used by Zone 1
Setup	Program	Setup	FXMods of program on channel assigned as FXChan	FX Chan
Program	Setup	Control setup	The last setup that was selected in Setup mode	Channel used by Zone 1 of the last setup
Program	Master	Control setup	Nothing	None
Setup	Master	Setup	Nothing	None

Table 15-2 Modes, Control Assignments, and Effects Control

Using KDFX With a Sequencer

If you use an external MIDI sequencer, you are probably thinking about how powerful it will be putting KDFX under sequencer control. Certainly being able to record, edit, and automate a studio’s parameters as part of a MIDI sequence is one of the most attractive aspects of KDFX.

FX Channel

Any program on any channel can be the one that controls KDFX. On the Effects-mode page, set FX Mode to **Program**, select the FX Channel you want to control KDFX with, and put your KDFX-controlling program on that MIDI channel. Now any MIDI commands coming from the sequencer on that MIDI channel will be sent to KDFX.

Dedicating a Program and Channel

Perhaps the most efficient and least confusing way to do this is to have a dedicated program that controls *only* KDFX, on a channel that is otherwise not being used to play music. Many K2500 and K2000 users know this trick for automating the old internal effects, but it becomes even more important given the complex nature of KDFX. It requires sacrificing a MIDI channel, but few users should have a problem with that.

Again, go to the Effects-mode page and, keeping the FX Mode set to **Program**, set the channel you’re going to use as an FX Chan—in this case, **15** is often a good choice. Go to Program Mode and select Program **199 Default Program**. Press **Edit**, and then **KEYMAP**, and set the Keymap parameter to a value of **0 None**. This program will now make no sound in response to MIDI notes, and use up none of the K2600's polyphony. Use the **<more>** buttons to get to the KDFX page, and start setting up your studio and FXMods, or, if you have another program already set up with the studio and FXMods you want, use Import FX to bring that studio and FXMods into the current program.

Now save the program to a new location, giving it a name like **Studio Controller1**. You can now use this studio with your sequencer: start by calling up its bank and program number, and then put appropriate MIDI commands into the sequencer for controlling the studio's parameters.

If the MIDI sources for the FXMods are also K2600 on-board controllers, then you can record your parameter changes from the K2600 into the sequencer.

There's a step-by-step discussion of this procedure, beginning on page 12-21.

Changing Studios With a Sequencer

If you need more than one studio available in a piece, or you want a selection of studios to use in different pieces, simply create new programs the same way, with different studios specified on their KDFX pages. To switch from one to the other, send an appropriate Program Change command on channel 16 from your sequencer.

Setting up static FXMods in different programs, all of which address the same studio, as we saw earlier, is a good way to make one-shot changes in KDFX parameters without having to construct a bunch of different studios.

Also as we saw earlier, this technique can be useful even when you are playing the K2600 in Setup mode along with a sequencer. If FX Mode is **Program**, and the FX channel is **16**, the sequencer can control the FXMods independently of which setups are in use on the keyboard.

Preventing Glitches When Changing Studios

As with any digital effects unit, you need to take some care when you are sending KDFX real-time commands that radically alter the nature of the processing it is doing. The trickiest situations will occur when you are changing studios, and calling up new FX presets, algorithms, and/or signal routings.

Under the best of circumstances, the transition between two studios can be seamless, and the effects in one will "morph" into the effects of the other. Under the worst of circumstances, there will be a momentary "hole" in the sound, as the effects from the first studio are cleared out and the effects of the second studio build up.

The chances of a smooth transition between studios will be highest if the algorithms on the FXBuses in the two studios are the same. (In some reverb algorithms, the Room Type parameter should also be the same.) For example, if FXBus1 in Studio X uses an FX preset based on the reverb in algorithm 4, and FXBus1 in Studio Y uses a different FX preset based on the same reverb algorithm 4, then when you switch from Studio X to Studio Y, the signals going through FXBus1 should experience a smooth reverb change.

However, if the FX preset for FXBus1 in Studio Y uses a *different* algorithm, say a multitap delay in Algorithm 35, then at the moment the studio changes, the reverb effect will ramp down quickly, there will be a very brief point at which the signal will pass dry, and then the multitap will quickly ramp up. This ramping will take place any time the two algorithms (or Room Types) are different—even if on the surface they seem very similar, like a Room and a Hall.

Even if the algorithms are the same, however, there is a chance that the transition will not be completely smooth. Some parameters cause glitches if they are changed in real time even within a studio—delay lengths, for example—and so if a parameter like that changes when you *switch* studios, the software will clear it out as the studio changes, and a small hole will appear. In some cases, instead of a hole, you will hear a momentary pitch shift as the studio changes. A bit of experimentation will help you determine how to achieve acceptable transitions between studios.

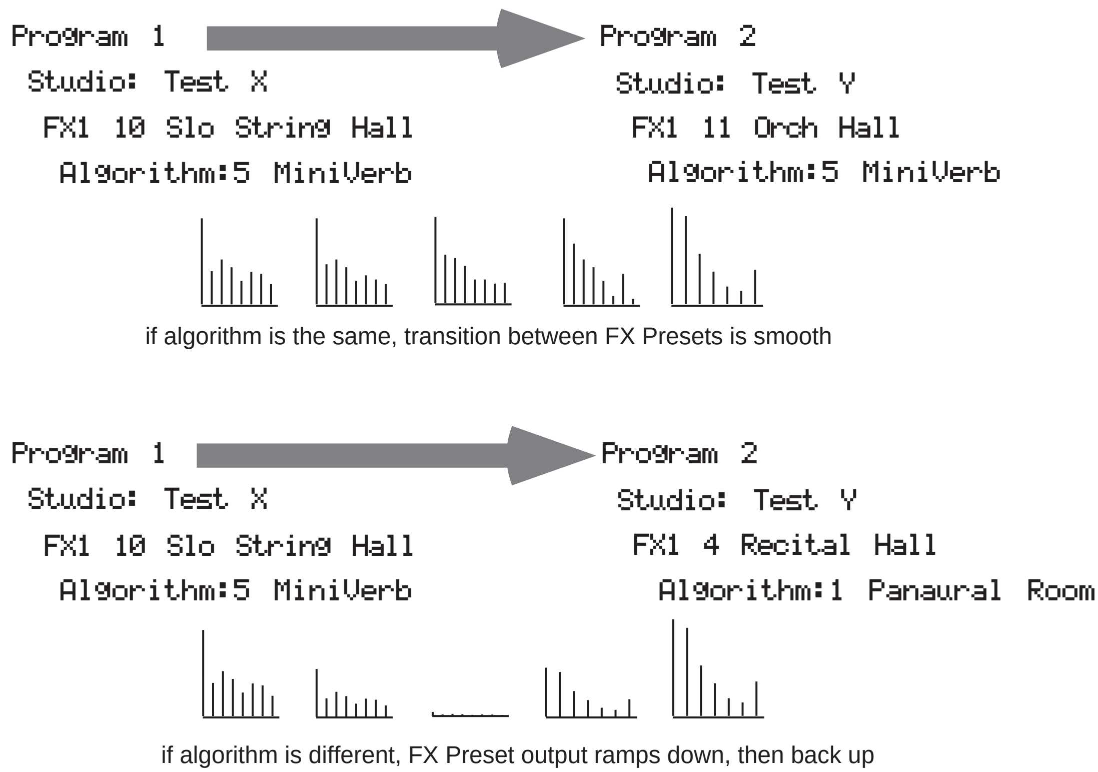


Figure 15-10 Effect of FX Algorithm on Transition Between FX Presets

Changing PAU Allocations

There are other circumstances under which using the same algorithm will not ensure a smooth transition. If for any reason the PAUs have to be reallocated when you move from one studio to another, then even if the algorithms on a given FXBus are the same in both studios, there will be a ramp-down/ramp-up. This can happen when there is a change in the number of PAUs being used on any FXBus that has a *lower number* than the FXBus you're concerned with. That's because reallocating PAUs on the fly forces some of the processors to reconfigure themselves, inasmuch as they are now being called upon to do different functions.

Here's an example: say Studio X's FXBus1 uses a Flanger that requires 2 PAUs, while FXBus2 has a small reverb that uses 1 PAU. In Studio Y, FXBus1 has a Chorus that uses only 1 PAU, while FXBus2 uses the same small reverb as Studio X. When you switch from Studio X to Studio Y, one of the PAUs that was previously being used for FXBus1 is now being used for FXBus2—it's been reallocated. Since it's not the *same* PAU that's handling the reverb, the transition isn't going to be smooth—even though the algorithm hasn't changed.

There are a couple of ways around this. First is to set up your studios so that any transitions between them that are going to force changes in PAU allocations occur in the *higher-numbered* FXBuses—if the situation just described were reversed, and the reverb was in FXBus1, there would be no problem, because that same PAU would be used in FXBus1 in both studios. A second method is to *manually* allocate the PAUs in the lower-numbered FXBuses, rather than using Auto allocation. Using the same example, if Studio Y had 2 PAUs hard-assigned to FXBus1, even though the FX preset is only using one of the PAUs, then that second PAU would not get reassigned to the reverb, and the reverb's transition would be smooth.

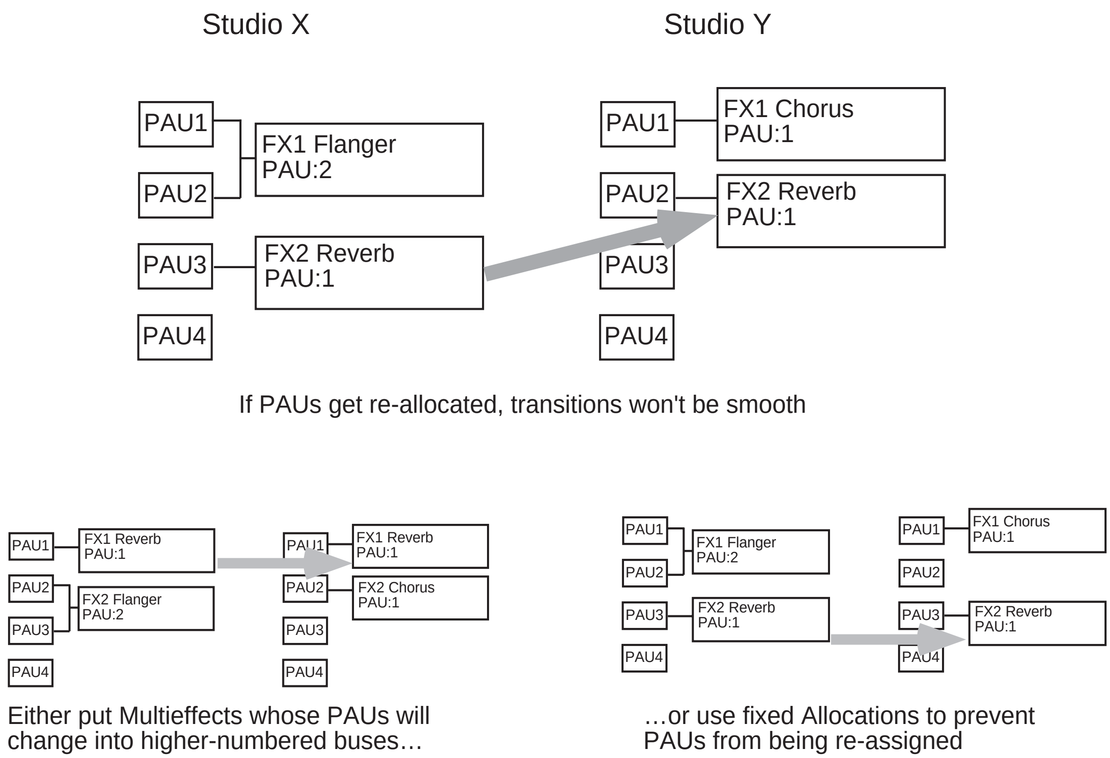


Figure 15-11 Avoiding Transition Problems When PAUs Get Reallocated

Tempo-based Control of KDFX

Many parameters within KDFX can be set up to respond to musical tempo. The tempo information can come from a number of sources: it can be fixed within a studio, it can come from the K2600's internal sequencer, or it can come from an external MIDI sequencer that has been configured to send MIDI Timing Clocks to the K2600.

The procedures for using tempo from the internal sequencer and from an external sequencer are essentially the same, with one crucial difference in one setting: in Song mode, on the MISC page. The Clock parameter must be set to **Int** when using the internal sequencer and to **Ext** when using an external sequencer.

Note that MIDI Timing Clocks (otherwise known as MIDI Sync) is not the same as MIDI Time Code (or MTC). MIDI Sync, which is used for syncing sequencers together, changes its speed in response to tempo changes; MIDI Time Code, which is used for syncing a sequencer to an audio or video tape or disk system, does not. KDFX (and the K2600 in general) responds to MIDI Sync, but not to MIDI Time Code.

Tempo Within an FX Preset

Some algorithms have a Tempo parameter, which allows certain time-based functions, such as LFOs, envelopes, or delay lengths, to be expressed in beats per minute (BPM). A second parameter immediately below Tempo, Period, multiplies or divides the tempo. The Period is expressed in beats (bts) and ranges from 1/24 (the actual time is 24 times faster than the Tempo) to 32 (the time is 32 times slower than the Tempo) in a sliding scale based on 24ths of a beat. So for example, if an LFO has a Tempo setting of 120 BPM and a Period of 12/24 bts, the LFO will oscillate 240 times per minute, or 4 Hz.

Variable Tempos Using System

You can also have these parameters respond to tempo in real time, by setting the Tempo parameter to **System**, which is set by scrolling below 1 BPM (or pressing **0** and **Enter** on the alphanumeric buttonpad). Now instead of Tempo being a fixed value, it will vary according to the system clock—which, depending on the setting of the Clock parameter on the Song mode's MISC page as described above, will either be the internal sequencer's clock, or tempo coming from an external source of MIDI Clock. The System Tempo will be multiplied or divided by the Rate or Beats setting, to determine the parameter's actual speed, just as if it were a fixed value.

If the Clock setting is **Int**, then the current tempo of the internal sequencer (as shown on the Song Mode's MAIN page) will be in effect regardless of whether the sequencer is running.

If the Clock setting is **Ext**, the external sequencer (assuming it is sending MIDI Clock) controls the Tempo parameter. If *no* MIDI Clock is being received by the K2600, the tempo will not go to **0**; instead, it will retain the last valid value it saw—either the tempo of the internal sequencer at the time when you switched the Clock to **Ext**, or the last tempo sent by an external sequencer before it stopped.

Tempo as an FXMod Source

KDFX algorithms that don't have a Tempo parameter can also respond to tempo information, using FXMods.

In the Source list, there is a value called Tempo, located at number 55. This source reads the current tempo (fixed or variable, internal or external) and turns it into a numerical value between **0** and **1**, which is then applied to the parameter under the control of the FXMod. The "virtual" range of this parameter is 3.75 to 240 BPM—that is, a tempo of 3.75 BPM (or less) will set the parameter value to **0**, while a tempo of 240 BPM (or greater) will set it to **1**. Of course, most parameters don't actually vary between **0** and **1**, so you have to scale the real values appropriately.

This can get pretty confusing, but there is a relatively simple method to follow when using this feature: set the Adjust value of an FXMod to the value of the parameter that you want when the tempo is 3.75 BPM, and set the Depth value so that the sum Adjust+Depth equals the value that you want when the tempo is 240 BPM.

For example, let's look at an algorithm that has an LFO Rate parameter in Hz. Let's say we want to the LFO speed to be twice as fast as the tempo. Here are the values for the FXMod:

Parameter	**LFO Rate**
Adjust	The slowest vibrato we want is (3.75 BPM/60) x 2, or 0.125 Hz. **0.13 Hz** is the nearest value available.
Source	**Tempo**
Depth	The fastest vibrato we want is (240BPM/60) x 2, or 8.00 Hz, so this parameter is 8.00–0.13, or 7.87 Hz; the nearest value available is **7.80 Hz**.

The Tempo value exists in Control Source lists throughout the K2600's operating system, not just in FXMod pages, so you can use it for controlling just about any function, including pitches of sample loops, envelope lengths, filters, volume, and more.

Tap Tempo

A real-time tap on a footswitch or button can also be used to control the tempo of a parameter in KDFX. Like Tempo, Tap Tempo is not KDFX-specific, but can be used throughout the K2600's operating system, including the sequencer.

Tap Tempo is on the Control Source list for any controller in a setup or control setup, located at number 39. Tapping or pressing on a controller that is set to **Tap Tempo** sends tempo information, based on the average time between taps, to the Internal Clock.

It takes four taps to determine the tempo—if you tap fewer than four times, no information is sent. If you tap more than four times, the average of *all* of your taps is continuously calculated and sent to the Clock. Later taps are weighted more heavily than earlier ones, so that if you change the rate at which you tap, the tempo information being sent will change accordingly, but there is a "flywheel" effect, and the Clock's tempo changes will lag somewhat behind the changes in your tapping rate.

In order for Tap Tempo to have any effect, the Clock parameter on the Song Mode:MISC page must be set to **Internal**.

KDFX in Master Mode

So far we've been telling you that setting FX Mode to **Master** on the Effects page disables real-time control over KDFX, but this is not strictly true. In that mode, KDFX will respond to specific system-exclusive (Sysex) commands. If you are a serious Sysex jockey, you can take advantage of this by using Sysex commands.

Sysex, in fact, can be used to control KDFX *regardless* of the FX Mode setting, so if you need to tweak certain parameters in real time without setting up an FXMod to do it, you can use Sysex and also use the FXMods in a program or setup.

See Appendix B in the *Musician's Reference* for details about using SysEx to control KDFX.

KDFX in Auto Mode

When the FX Mode is set to **Auto**, then control of KDFX changes according to where you take the K2600: either Program mode or Setup mode. But when you are on the Effects-mode page itself, you are actually in *Master* mode. Therefore, you can go into the studio whose name appears on the page and tweak its parameters, but there won't be any FXMods. You can change a parameter that has an FXMod link through a program (or setup), but, unlike when you are going into the studio through a program, that link will *not* be disturbed.

When you leave the studio, and then go into Program or Setup mode and call up a program or setup that links to that studio, any changes you have just made in the studio will show up, and any FXMods previously in place will also still be there.

From Auto Mode to Song Mode

When you are on the Effects-mode page and if FX Mode is set to **Auto**, and then you go into Song Mode, something slightly different happens: KDFX is now under control of the program that is on the effect channel within the song—from Song mode press **Edit** and look at the EffectChan parameter.

The effect channel is saved with the song, so make sure that it is set so that the program that you want to be in control of KDFX is active.

Using the Algorithms

This section will discuss the processing algorithms found in KDFX: what their parameters mean, and how to use them. It is designed to be read through so that you can get a general understanding of the algorithms. You can find a more complete reference, with every algorithm and the meaning and range of every parameter, arranged in the order they appear on the screen, in the *Musician's Reference*.

But first, a word about how the KDFX studios in ROM are organized.

Conventional Studio Structure

Many of the studios provided in ROM follow an overall organizational plan, which uses KDFX's resources efficiently and clearly. While by no means are you required to follow this structure when creating your own studios, it's a good idea to become familiar with it, and see how Kurzweil's own engineers have approached the complex issue of studio organization. And certainly it can serve as a good starting point when you start creating your own studios.

The organization of the ROM studios follows the following guidelines Not all studios are set up this way, but most of them are.

- ROM programs and setups are assigned to an output (KDFX-A through KDFX-D on the Program:Output or Setup:CH/PRG page) based on the type of effects processing that would most likely be appropriate for that sound.
- All inputs go to their respective FXBuses (Input A to FXBus 1, Input B to FXBus 2, etc.), and *only* to their respective FXBuses—the Lvl parameter for the second FXBus on each Input page is turned Off.
- FXBus1 contains a relatively simple reverb with a low Size requirement.
- FXBus2 contains an effect that does not increase the "length" of the sound (that is, no reverb or delay). Possibilities include chorus, flange, phaser, distortion, shaper, pitcher, enhancer, EQ, or EQ morpher.
- FXBus3 contains effects that take up lots of time, such as delays, delays with reverb, or other lead sounds.
- FXBus4 is dry, since the first three FXBuses have probably used up all the PAUs.
- The Aux bus contains a larger reverb (Size: 2 or 3), a compressor, or a graphic EQ. It can often be used instead of an FXBus reverb, such as the one on FXBus1. If you use it in this way (set the Aux Lvl on FXBus1 to 0dB or higher), it frees up FXBus1 for use as an Enhancer, Stereo Image, Flanger, etc.

General Parameters

Some parameters show up in all algorithms, and we'll deal with those first.

Wet/Dry balances the levels of the processed and unprocessed signals passing through the FXpreset. The range is 0% wet (the signal is unprocessed) through 100% wet (no dry signal is present). A setting of 50% wet means the dry and processed signals are equal in level. In some algorithms, separate Wet/Dry parameters are provided for the Left and Right input channels.

In Gain sets the level of the signal coming into the FXpreset from the Input page. As with most Gain controls in KDFX, the range is -79.0 dB to 24.0 dB, and there is an **Off** position. **0.0 dB** is unity gain.

Out Gain sets the level of the signal after it passes through the FXpreset. From here the signal goes directly to the Output page, if one of the Outputs is set to FXBus*n*. The level can be further changed before it goes to the Mix and/or Aux buses.

In/Out enables or disables the effect. It's like a Wet/Dry parameter with only two positions: 100% (In) and 0% (Out).

HF Damping is the cutoff (-3 dB) frequency of a 6dB/octave lowpass filter that's inserted before the processor. In the case of processors where multiple iterations of the signal are heard, such as in a delay, each iteration of the signal will pass through the filter, and will therefore be duller.

XCouple (Cross Couple). In stereo effects, this controls how much of any signal that is being fed back goes to the channel opposite to the one where it first appeared. At 100%, all feedback from signals at the left input goes to the right channel, and vice versa, causing a "spreading" or in the case of delay lines, a ping-pong effect. At 0%, fed-back signals stay with the channel they came in on.

A->B cfg (configuration). In combination algorithms that contain two components, and whose name uses **<>**, as in **713 Flange<>Shaper**, the order in which the signal passes through the two components can be changed. For example, this algorithm can be configured so the signal passes through the reverb first and then the compressor, or through the compressor first and then the reverb. The A->B cfg parameter determines the configuration, and its value is context-sensitive—in this example, the choices would be **Rvb->Cmp** and **Cmp->Rvb**.

A/Dry->B is also found in many combination algorithms, and controls the amount of signal that will pass dry (unprocessed) through the first component into the second component. Different combination algorithms use different variations on this parameter, depending on the context. The range is 0 to 100%.

Reverbs

Room Type changes the configuration of the algorithm to simulate a wide array of room types and sizes including booths, small rooms, chambers, halls and large spaces can be selected. Use this parameter to choose different types of reverb—if all other parameters remain at their nominal values, what you get when you change this parameter will always sound great. Because this parameter changes the structure of the reverb algorithm, it cannot be assigned an FXMod. Room types in different algorithms with similar names do not necessarily sound the same.

Rvrb Time is the RT_{60}—the time it takes for the reverb to decay to 60 dB below its initial level—in seconds. It is accurate assuming that several other parameters (HF Damping, Diff Scale, Size Scale, and Density) are at their nominal levels. It is adjustable up to Inf, which creates an infinitely-sustaining reverb.

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

L/R Pre Dly (PreDelay) is the time 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.

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.

Diff Scale scales the diffusion of the early reflections, that is, how spread out they are as a group over time. At very low settings, the early reflections start to sound quite discrete, and at higher settings the early reflections are seamless. It is adjustable from 0.00 to 2.00, with 1.00 being nominal for the given Room Type.

Density controls how tightly the early reflections are packed in time. Low Density settings group the early reflections close together, while higher values spread the reflections for a smoother reverb. It is adjustable from 0.00 to 4.00, with 1.00 being nominal (and usually optimal) for the given Room Type.

Expanse controls the 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.

Build adjusts the envelope of certain portions of the reverb. Positive values speed up the envelope, and negative values slow it down.

Size Scale changes the size of the current room. Altering this parameter will cause a subtle coloration of the reverb. It is adjustable from 0.00 to 4.00, with 1.00 being nominal (and usually optimal) for the given Room Type.

InfinDecay, when set to **On**, causes the reverb tail to decay indefinitely. When it's **Off**, the decay time is determined by the Rvrb Time or LateRvbTim parameters.

Wet Bal (Wet Balance). Some reverb algorithms are actually two stereo reverbs in one, with each one receiving a different mono signal. This balances the outputs of the two reverbs—0% means they are being mixed equally.

Delays

There are two types of taps in the Multitap delays: The Loop tap, which can be repeated, and the numbered taps, which play a single iteration.

Fdbk (Feedback) **Level** controls the repeating function of the Loop Tap. A setting of **0**% means there will only be a single delay, while a setting of **100**% means the signal keeps repeating without ever stopping.

Both types of taps are individually adjustable from 0 to 2.55 seconds. The Loop Crs and Tap*n* Crs (*n* being the number of the tap) parameters set the coarse value of the loop in 20-ms increments, while the Loop Fine and Tapn Fine parameters set the fine value in 0.2-ms increments.

In Delay algorithms that use tempo to determine tap lengths, there is a Tempo parameter, which can be set from **1** to **255** BPM or to **System**. The Loop Length and Tap*n* Delays are then expressed in beats relative to that overall Tempo. See *Tempo-based Parameters* on page 15-25 for more information about tempo control of KDFX parameters.

Hold is a switch that, when turned on, "locks" any signal currently in the delay and plays it until Hold is turned off. When Hold is on, no signal can enter the delay and Feedback is set to 100%.

Dry Bal (Balance) is the left/right balance of the dry signal. At -**100**%, only the left dry signal goes to the left output, while at **100**% only the right dry signal passes to the right output, and at **0**%, equal amounts of the left and right dry signals pass to their respective outputs.

Tap*n* Level is the level of each numbered tap, from **0**% to **100**%, relative to the overall output of the effect.

Tap*n* Bal is the left/right balance of each of the numbered taps. At **-100**%, only the left tap goes to the left output, while at **100**% only the right tap goes to the right output, and at **0**%, equal amounts of the left and right taps pass to their respective outputs. In some delays, pairs of taps (1 and 5, 2 and 6, etc.) are controlled together as stereo pairs.

DelayScale lets you change the lengths of all the taps together. Its range is **0** to **10x**.

Note that it is possible for KDFX to run out of delay memory with over-generous settings of DelayScale or very slow Tempos. If this happens, delay times will be automatically cut in half.

Complex Echo

This algorithm has two feedback taps per channel as well as three independent taps, and also a feedback diffuser for “smearing” the delays. 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.

FB2/FB1>FB is a balance control between feedback lines 1 and 2. **0**% (minimum) turns off feedback line 2, only allowing use of feedback line 1. **50**% is an even mix of both lines, and **100**% (maximum) turns off line 1.

L Diff Dly and **R Diff Dly** adjusts the delay lengths of the diffusers. Range is **0** to **100 ms**.

Diff Amt adjusts the diffuser intensity. Range is **0** to **100**%.

***N* Fdbk*n* Dly** adjusts the delay length of the *N* channel’s *n*th feedback tap, fed back to the *N* channel’s delay input. Range is **0** to **2600 ms**.

Spectral Multitap Delays

These 4- and 6-tap delays have their feedback and output taps modified with shapers and filters. In the feedback path of each tap are a diffuser, hipass filter, lopass filter, and imager. Each delay tap has a shaper, comb filter, and balance and level controls.

Fdbk Image sets the amount the stereo image is shifted each time it passes through the feedback line. Range is **-100** to **100**%.

Tap *n* Shapr adjusts the intensity of the shaper at each output tap. Range is **0.10** to **6.00 x**.

Tap *n* Pitch adjusts the frequency of the comb filter at each output tap. Range is **C -1** to **C 8**, in semitones.

Tap *n* PtAmt adjusts the intensity of the comb filter at each output tap. Range is **0** to **100**%.

Equalizers (EQ)

KDFX has both Graphic and Parametric EQ algorithms. Parametric EQ sections are also available on a number of combination algorithms.

The Graphic equalizer is available as stereo (linked parameters for left and right) or dual mono (independent controls for left and right). It has 10 bandpass filters per channel, each of whose gain is adjustable from **-12 dB** to **+24 dB**.

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. To minimize this ripple, it is best to center the overall settings around **0 dB**.

The Parametric equalizer (5-Band EQ) has two bands of shelving filters and three bands of true parametric EQ.

Treb Freq and **Bass Freq** set the center frequencies for the shelving filters. Both of these are adjustable over the full range of **16** to **25088 Hz**, in increments of a semitone.

Treb Gain and **Bass Gain** control the amount of cut or boost above (Treb) or below (Bass) the center frequency. The range is **-79** to **+24 dB**.

Mid*n* Gain sets the cut or boost for the parametric band *n*, with a range of **-79** to **+24 dB**.

Mid*n* Freq sets the center frequency for parametric band *n*, with a range of **16** to **25088 Hz**, in increments of a semitone.

Mid*n* Width sets the bandwidth of the filter on band *n*, with a range of **0.01** to **5 octaves**.

Enhancers

Enhancers modify the spectral content of the input signal by boosting existing spectral content, or stimulating new ones. Two- and three-band versions are provided.

Drive adjusts the input into each band. Increasing the drive will increase the effects. Range is **-79.0** to 2**4.0 dB**.

Xfer adjusts the intensity of the transfer curves. Range is **-100** to **100**%.

EQ Morpher

This algorithm uses two four-band bandpass filters, A and B, and moves between them, which among other things, can produce a very convincing simulation of a human vocal tract.

FreqScale offsets the filter frequencies for each set of filters. After setting the filter parameters (Freq, Gain, and Width), the FreqScale parameters will move each of the four filter frequencies together by the same relative pitch. Range is **-8600** to **8600 cents**.

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. Range is **0** to **100**%.

Compressors, Expanders, and Gates

A wide range of Compression and Expansion effects is available in KDFX. The various algorithms include different combinations of:

- Compressor with either soft-knee or hard-knee characteristic
- Expander
- Multiband compressor that breaks the signal up into three frequency bands and compresses them all separately
- Sidechain or output EQ
- Reverb and compressor in combination
- Gate
- Gated reverb

All of the compression algorithms use these parameters:

FdbkComprs (Feedback Compression) selects whether to use feed-forward (set this to **Out**) or feedback (set this to **In**) compression. The feed-forward configuration uses the input signal as the side-chain source. The feedback configuration uses the compressor *output* as the side-chain source. Feedback compression tends to be more subtle, but it does not allow instantaneous attack compression.

Atk (Attack) **Time** for the compressor is adjustable from **0.0** to **228.0 ms**.

Rel (Release) **Time** for the compressor is adjustable from **0** to **3000 ms**.

Smooth Time smooths the output of the expander's envelope detector by putting a lowpass filter in the control signal path. Smoothing will affect the Attack or Release times only when this parameter is longer than one of the other times. The range is **0.0** to **228.0 ms**.

Signal Dly (Delay) puts a small delay in the signal relative to the sidechain processing, so that the compressor (or gate) "knows" what the input signal is going to be before it has to act on it. This means the compression can kick in before an attack transient arrives. In a compressor, it is only really useful in feedback configuration (FdbkComprs is **In**). The range is **0** to **25 ms**.

Ratio is the amount of gain reduction imposed on the compressed signal, adjustable from **1.0:1** (no reduction) to **100:1**, and **Inf:1**.

Threshold is the level in dBFS (decibels relative to full scale) above which the signal begins to be compressed. Adjustable from **-79.0** to **0 dB**.

MakeUpGain allows additional output gain to compensate for gain reduction in the compressor. It works in conjunction (additive, in dB) with the Out Gain parameter. The range is **-79.0** to **+24.0 dB**.

Expansion

Algorithms containing Expanders have these controls:

Exp Atk (Attack), how fast the expander turns off when the input signal rises above the threshold level, adjustable from **0.0** to **228.0** ms.

Exp Rel (Release), how fast the expander turns back on after the signal drops below the threshold level, adjustable from **0** to **3000 ms**.

Exp Ratio, how much the gain is reduced below the expansion threshold, adjustable from **1:1.1** (slight downward expansion) to **1:17** (extreme downward expansion).

Exp Threshold, the level below which the signal is expanded, adjustable from **-79.0** to **0 dB**.

In addition, the two-segment compressors with expander have separate Ratio and Threshold controls for each of the compression segments.

Multiband Compression

The Multiband Compression algorithm has Attack, Release, Smooth, Signal Delay, Ratio, Threshold, and MakeUp Gain parameters for each of the three bands (Low, Mid, and High). In addition, it has the following:

Crossover1 and **Crossover2**. These set the frequencies which divide the three compression frequency bands. The two parameters are interchangeable, so either may contain the higher frequency value. The range is **16** to **25088 Hz**, in increments of a semitone.

Gates

SC Input lets you select which input channel(s) will control the sidechain, which is responsible for opening and closing the gate. It can be set to **L**, **R**, or the average of the two channels, **(L+R)/2**.

Gate Time is the time that the gate will stay open when the sidechain signal reaches the Threshold. Its range is **0** to **3000ms**.

Ducking reverses the action of the gate. Normally this if set to **Off**, and the gate opens when the input signal rises above the threshold. But when this is **On**, the gate *closes* when the input signal rises above the threshold.

Env Time is the amount of time it takes for the sidechain signal envelope to drop below the threshold. If this time is too short, the gate can close and open too quickly from amplitude modulation in the sidechain signal. If it is too long, the gate may stay closed until the envelope has a chance to fall, and some signals would not get through. This parameter is only in effect when Retrigger is **Off**.

Retrigger determines whether the gate timer will reset itself each time the sidechain signal goes above the threshold. If it is **On**, the timer resets itself, and therefore the gate stays open as long as the signal is above the threshold, or keeps going above the threshold, within the interval specified by Gate Time. If it is **Off**, the gate closes down after Env Time has elapsed—regardless of the sidechain level—and the sidechain level must fall below the threshold and come back up again before the gate will open again.

Chorus

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 to 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, while the dual mono choruses have separate left and right controls.

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

Tap Lvl sets the levels of the LFO-modulated delay taps. Negative values polarity-invert the signal. Setting any tap level to **0**% effectively turns it off.

Tap Pan sets the stereo position for a given tap’s output. The range is **-100**% for fully left to **100**% for fully right.

LFO Rate sets the speed of modulation of the delay lines with a range of **0.01** to **10 Hz**.

LFO Dpth sets the maximum detuning depth of the LFO-modulated delay lines, with a range from **0** to **50 cents** (1/2 semitone).

Tap Dly adds extra delay in front of the LFO modulated delay taps from **0** to **230 ms**.

L/R Phase or **LFO*n* LRPhs** adjusts the relative phases of the LFOs for the left and right channels in the stereo Choruses.

Flanger

Flanging is the process of adding or subtracting a signal with a time-displaced replica of itself, which results in a series of notches in the frequency spectrum, generally referred to as a comb filter. In KDFX, the flanger is a multi-tap delay line, all (but one) of whose taps can have their lengths modulated up and down by a low frequency oscillator (LFO). The rate of the LFO is expressed in Tempo.

StatDlyLvl (Static Delay Level) is the level of the first, nonmoving tap. Negative values invert the polarity of the tap. The range is **-100** to **100**%; **0**% turns the tap off.

DlyCrs and **DlyFin** are the coarse and fine length controls for the Static delay (StatDly…) and for the minimum value of the moving delays (Dlyn…). The coarse range is **0** to **228 ms**, and the fine range adjusts the coarse range in samples (1/48,000 sec, or 20.8μsec) from **-127** to **127**.

Xcurs Crs and **Xcrs Fin** determine how far the LFO-modulated delay taps can move from the center of their ranges (this is called *excursion*). 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 coarse range is **0** to **228 ms**; the range **0** to **5 ms** is most effective for flanging. The fine range adjusts the coarse range in samples from **-127** to **127**.

Quantize + Flange

The Quantize portion of this algorithm produces digital distortion known as quantization noise by limiting the number of bits available to the signal.

DynamRange (dynamic range) controls 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, thereby producing a square wave. Every 6 dB added doubles the number of quantization levels, reducing the noise and getting closer to the original signal. If the signal has a lot of headroom (available signal level before digital clipping), then not all quantization levels will be reached. Range is **0** to **144 dB**.

Headroom sets the available signal level before digital clipping. Use this in conjunction with DynamRange to keep the signal level from getting too loud at low levels of DynamRange. Range is **0** to **144 dB**.

DC Offset adds a positive DC Offset to the input signal, which allows you to alter the position where digital zero is with respect to your signal. At low DynamRange settings, this can cause the output to sputter. Range is **Off/-79.0** to **0.0 dB**.

LaserVerb

LaserVerb is a new kind of reverb which produces a delayed train of closely spaced reflections, or impulses. As time passes, the spacing between the impulses gets wider, which creates a discernible buzzy pitch that gets lower as the spacing increases. The signal can be fed back into itself to extend the effect.

Dly Coarse is the overall delay length, which controls the duration or decay time. 0.5 sec is a good starting point. Range is **0** to **1.3 seconds** in the 2 PAU version of the algorithm, and **0** to **2 seconds** in the 3-PAU version.

Dly Fine adjusts the delay with a resolution down to 0.1 ms. Range is **-20.0** to **20.0 ms**.

Spacing determines the starting pitch of the descending buzz and how fast it descends, by setting the initial separation of impulses and the subsequent rate of increasing impulse separation. The spacing between impulses is given in samples (20.8μs), with a resolution of 0.1

sample. At low values, the buzz starts at high frequencies and drops slowly, while at high values the buzz starts at a lower pitch and drops rapidly. Range is **0.0** to **40.0 samples**.

Contour controls the overall 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 more time for the effect to build up before decaying. At a value of around **34**, the reverb behaves like a reverse reverb, building up to a hit. When it is set to **0**, the algorithm acts like a simple delay. Range is **0** to **100**%.

Filters

There are four types of Resonant Filter algorithms in KDFX.

Resonant Filter

Filter Type (**FiltType**) can be **Lowpass**, **Highpass**, **Bandpass**, or **Notch** (band-limit).

Frequency (**Freq**) is the resonant frequency of the filter. Its range is **58** to **8372 Hz**.

Resonance is the resonance of the filter, adjustable from **0** to **50 dB**.

Envelope Filter

Envelope Filter is a resonant filter whose center frequency can be made to vary according to the level of the incoming signal.

Filter Type can be **Lowpass**, **Highpass**, **Bandpass**, or **Notch** (band-limit).

Min Freq is the minimum resonant frequency of the filter, that is, the filter frequency when the input gain is below the triggering threshold. Its range is **58** to **8372 Hz**.

Sweep determines how far the resonant frequency moves when the input level increases. At positive levels it moves up in pitch, and at negative levels it moves down. The highest possible resonant frequency is 8372 Hz,; the lowest is 0 Hz. This parameter's range is **-100**% to **+100**%.

Resonance is the resonance of the filter, adjustable from **0** to **50 dB**.

Atk Rate adjusts the upward slew of the attack portion of the envelope detector. Range is **0** to **300.0 dB/sec**.

Rel Rate adjusts the downward slew of the release portion. Range is **0** to **300.0 dB/sec**.

Smooth Rate slows down the envelope follower. If it is set to a lower rate than Atk Rate or Rel Rate, it can dominate those parameters. Range is **0** to **300.0 dB/sec**.

Trig Filt

The Triggered Filter is a sweeping resonant filter that triggers when a certain input threshold is reached, and then follows its *own* envelope, consisting of an instantaneous attack and an exponential release, rather than the envelope of the input signal.

Max Freq is the resonant frequency of the filter at the peak of the internal envelope. It can be set lower than Min Freq (above), in which case the filter will sweep downwards, then back up. Range is **58** to **8372 Hz**.

Trigger is the input-signal threshold at which the envelope detector triggers. Range is **-79** to **0 dB**.

Retrigger is the input-signal threshold at which the envelope detector resets, so that it can trigger again. This parameter is only useful when it is set below the value of Trigger. Range is from **-79** to **0 dB**.

Env Rate is the envelope detector decay rate. This 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 range is **0** to **300.0 dB/sec**.

Rel Rate is the downward slew (release) rate of the triggered envelope generator. The range is **0** to **300.0 dB/sec**.

Smth Rate slows down the envelope follower. If set lower than the release rate, it will dominate it. You can also use the smoothing rate to lengthen the attack of the internal envelope. The range is **0** to **300.0 dB/sec**.

LFO Filter

The LFO filter is continuously swept between two resonant frequencies over a period of time. The LFO frequency, expressed in BPM and beats, can be fixed or set to follow System tempo. (See *Tempo-based Parameters* on page 15-25for more information about tempo control of KDFX parameters.)

Min Freq and **Max Freq** are the low and high limits of the resonant frequency as the filter is swept. It actually doesn't matter which is higher; the effect will be the same. The range for both is **58** to **8372 Hz**.

LFO Shape is the waveform type for the LFO. Choices are **Sine**, **Saw+**, **Saw-**, **Pulse**, and **Tri**.

LFO PlsWid (Pulse Width). When the LFO Shape is set to **Pulse**, this sets the pulse width as a percentage of the waveform period. When the width is set to **50%**, the result is a square wave. This parameter has no effect if other waveform types are chosen. Range is **0** to **100%**.

LFO Smooth smooths (removes the higher harmonics from) the Saw+, Saw-, and Pulse waveforms. A Sawtooth wave looks more like a triangle wave, and a Pulse wave looks more like a sine wave. Range is **0** to **100%**.

Distortion

Distortion algorithms on KDFX may include a parametric equalizer or a cabinet simulator.

Dist Drive applies a boost to the input signal to overdrive the distortion algorithm into soft clipping. Since distortion drive will make your signal very loud, you may have to reduce the Out Gain as the drive is increased. Range is **0** to **96 dB**.

Warmth is 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. Range is **16** to **25088 Hz**.

Highpass allows you to reduce the bass content of the distortion content in the smaller distortion algorithms that don't have true parametric EQ. Range is **16** to **25088 Hz**.

Cab Preset selects from eight cabinet simulations which 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**.

Cab Bypass switches on and off the cabinet-simulation part of the algorithm. When this is set to **In**, the cabinet simulation is active; when it is **Out**, there is no cabinet filtering.

Cabinet HP and **Cabinet LP** are highpass and lowpass filters to set the frequency response limits of the cabinets. Range of both filters is **16** to **25088 Hz**.

Polydistort

This is a more complex distortion algorithm that provides two, four, or six stages of distortion.

Curve *n* controls 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. Maximum value is **127**%.

LP *n* Freq are shelving frequencies for one-pole lowpass filters on each of the distortion stages. LP0 Freq handles the initial low pass prior to the first distortion stage. The other low pass controls follow their respective distortion stages. Range is **16** to **25088 Hz**.

Rotating Speakers

An algorithm that includes Rotating Speakers breaks the signal into two frequency bands, “rotates” each band separately through a virtual speaker, and then combines the outputs with a pair of virtual “microphones” whose angle relative to the speakers is adjustable.

Xover (Crossover) is the frequency at which high and low frequency bands are split and sent to separate rotating drivers. The range is **16** to **25088 Hz**.

Lo Gain and **Hi Gain** are the gains of the signal passing through the rotating woofer or tweeter, respectively. The range is **Off**/**-79.0** to **24.0 dB**.

Lo Rate and **Hi Rate** are the rotation rates of the rotating woofer and tweeter. Each driver woofer can rotate clockwise or counter-clockwise, which is determined by the sign of this parameter: assuming the microphones are positioned in front of the driver and the microphones are panned positively (positive numbers go to the right), then a positive value for this parameter causes the driver to spin clockwise when viewed from the top. The range is **-10.00** to **10.00 Hz**.

Lo Size and **Hi Size** are the effective sizes (radius of rotation) of the rotating speakers in millimeters. This affects the amount of Doppler shift or vibrato of the low frequency signal. The range is **0** to **250 mm**.

Lo Trem and **Hi Trem** control the depth of tremolo (amplitude modulation) of the signals. It is expressed as a percentage of full scale tremolo. The range is **0** to **100**%.

Mic Angle is 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. You should not assign an FXMod to this parameter because adjustments to it will result in large sample skips, which will cause clicks in the signal passing through. The range is **0** to **360.0 degrees**.

LoResonate and **HiResonate** are simulations of cabinet resonant modes express as a percentage. For realism, you should use very low settings. The range is **0** to **100**%.

Lo Res Dly and **Hi Res Delay** are the number of samples of delay in each resonator circuit in addition to the rotation excursion delay. The range is **10** to **2550 samples**.

LoResXcurs and **HiResXcur**s are the number of samples of delay to sweep through the resonator at the rotation rate of each rotating speaker. The range is **0** to **510 samples**.

ResH/LPhs 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. The range is **0** to **360.0 degrees**.

Vibrato/Chorus

The Vibrato/Chorus algorithm (and also the KB3 effects algorithm) simulates the vibrato and chorus effects on a Hammond organ, and is used in conjunction with the Rotary Speaker. It has several unique parameters:

VibChInOut is an in/out switch for the Vibrato/Chorus effect.

Vib/Chor is the type of Vibrato/Chorus effect to be used. The choices are from three vibratos, **V1**, **V2**, **V3**, or three choruses, **C1**, **C2**, **C3**.

Roto InOut engages or bypasses the rotary speaker effect.

Lo Beam W and **Hi Beam W** set the acoustic radiation patterns (beam width) of the two drivers in the rotating speaker. If you imagine looking down on the rotating speaker, this angle is the angle between the -6 dB levels of the beam. The range is from **45°** to **360°**. At **360°**, the driver is omnidirectional.

There are four virtual microphones, with two each on the woofer (**LoMic A** and **LoMic B**) and on the tweeter (**HiMic A** and **HiMic B**). Each microphone has the following parameters:

- **Pos** (position), the angle of the microphone from the front of the virtual speaker, from **-180** to **180 degrees**
- **Lvl** (level) from 0 to 100%
- **Pan**, the left/right panning of the microphone's output, from -100% (full left) to 100% (full right)

Tremolo and AutoPan

Tremolo is amplitude modulation using an LFO. AutoPan moves the signal between the left and right channels, using an LFO. They have several parameters in common and several unique parameters.

LFO Rate is the rate of the LFO. The range is **0** to **10.00 Hz**, or in Tremolo BPM algorithm, **0** to **12.00 x** the tempo.

Rate Scale multiplies the speed of the LFO rate into the audio range. The range is **1** to **25088 x**. When above **19x**, the values increment in semitone steps. These steps are accurate when LFO Rate is set to **1.00 Hz**.

LFO Shape is the waveform type for the LFO. Choices are **Sine**, **Saw+**, **Saw-**, **Pulse**, and **Tri**.

LFO PlsWid or **Pulse Width**: When the LFO Shape is set to **Pulse**, this sets the pulse width as a percentage of the waveform period. When the width is set to **50%**, the result is a square wave. This parameter has no effect if other waveform types are chosen. Range is **0** to **100%**.

Origin (AutoPan) determines the axis for the panning motion. At **0%**, the panning is centered between the speakers. Positive values shift the axis to the right, while negative values shift it to the left. At **-100%** or **+100%** (the range limits), there is no panning action.

ImageWidth (AutoPan) is the width of the original input program material before it is auto-panned. At **0%** (minimum), the input image is shrunk to a single point source, allowing maximum panning excursion. At **100%** (maximum), the original width is maintained so no panning can occur.

Pan Width (AutoPan) controls the amount of pan excursion. It is the percentage of total panning motion available after Origin and ImageWidth are set. Range is **0** to **100%**.

CentrAtten (Attenuation) (AutoPan) is the amount the signal level drops as it is panned through the center of the stereo image. 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, while values below **-3dB** will cause a dip. Range is **-12** to **0 dB**.

Depth (Tremolo) controls the amount of attenuation applied when the LFO is at its deepest excursion point. Range is **0** to **100**%.

LFO Phase (Tremolo BPM) shifts the phase of the tremolo LFO relative to the beat reference. It is most useful when Tempo is set to **System**. Range is **0.0** to **360.0 degrees**.

50% Weight (Tremolo) is the relative amount of attenuation added when the LFO is at the -6dB point. This causes the LFO shape to bow up (positive values) or down (negative values). Range is **-16** to **3 dB**.

L/R Phase sets the phase relationship of the channels. **In** flips the left channel's LFO out of phase, with the result that the effect turns into an auto-balancer. **Out** leaves the left LFO alone.

Pitch Shifter (Pitcher)

Pitcher shifts the pitch of the incoming signal to the specified note.

Pitch is the fundamental pitch imposed upon the input, in MIDI note numbers from **C -1** to **G 9**.

Ptch Offst is an offset from the pitch frequency in semitones, from **-12.0** to **12.0**. It can be useful to assign pitch bend or a continuous controller to this parameter through an FXMod.

Odd Wts, **Pair Wts**, **Quartr Wts**, **Half Wts** are parameters that 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 more information and examples, see the algorithm documentation.

Ring Modulation

Ring modulation multiplies two signals (the "carrier" and the "modulator") together to produce unusual, often nonharmonic, overtones. The Ring Modulator algorithm in KDFX has two modes: **L*R**, in which two mono signals are modulated together; and **Osc**, in which the input is stereo, and it is modulated with the sum of five waveforms which are generated within the algorithm itself. Four of these are sine waves, and one (Oscillator 1) has a configurable waveform.

Wet/Dry: In L*R mode, the left signal is passed dry through this control.

Mod Mode selects between the two modes.

Osc1 Lvl is the level of Oscillator 1, from **0** to **100**%.

Osc1 Freq is the frequency of Oscillator 1, from **16** to **25088 Hz**.

Osc1 Shape is the waveshape of Oscillator 1, selectable from **Sine**, **Saw+**, **Saw-**, **Pulse**, and **Tri**.

Osc1PlsWid (Pulse Width). When Osc1 Shape is set to **Pulse**, this sets the pulse width as a percentage of the waveform period. When the width is set to **50**%, the result is a square wave. This parameter has no effect if other waveform types are chosen. Range is **0** to **100**%.

Osc1Smooth smooths (removes the higher harmonics from) the Saw+, Saw-, and Pulse waveforms. A Sawtooth wave looks more like a triangle wave, and a Pulse wave looks more like a sine wave. Range is **0** to **100**%.

The other four oscillators, **Sine2** through **Sine5**, each have Lvl and Freq controls.

SRS (Sound Retrieval System)

SRS, which is used under license from SRS Labs, Inc., is a single-ended processing system that produces a fully immersive, three-dimensional sound image from any audio source—mono, stereo, surround sound or encoded with any other audio enhancement technology—with two or more standard speakers.

The four 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. All ranges are **-79.0** to **24.0 dB**.

Center varies the amount of “center channel” in the output. It has an **Off** position.

Space controls the width of the image. It, too, has an **Off** position.

Bass Gain and **Treb Gain** set the amount of ambience added to the low and high frequencies, respectively, in the signal. Setting both of these to **0 dB** gives a best match to the optimizations of SRS Labs.

Stereo Simulation

The Mono to Stereo algorithm converts a monaural input to simulated stereo output.

In Select selects the input signal to be “stereoized”. It can be **Left**, **Right**, or both (**(L+R)/2)**).

CenterGain is the level of the summed left and right channels. Range is **Off**/**-79.0** to **24.0 dB**.

Diff Gain is the level of the difference signal produced, which is the spatial component of the stereo signal. Range is **Off**/**-79.0** to **24.0 dB**.

DiffBassG is a gain parameter for a bass-shelf filter on the difference signal. By boosting the low frequency components of the difference signal, you can increase the sense of acoustic envelopment. Range is **-79.0** to **24.0 dB**.

DiffBassF is the transition frequency for the bass-shelf frequency. Range is **16** to **25088 Hz**.

The processed signal is split into three frequency bands—Lo, Mid, and High—each of which can be delayed and panned separately.

Crossover1 and **Crossover2** are the two Crossover frequencies at which the band-split filters split the signal into three bands. The two parameters are interchangeable: either may have a higher frequency than the other. Range is **16** to **25088 Hz**.

Pan band sets the pan position for each band. Range is **-100**% (fully left) to **100**% (fully right).

Delay band sets the delay for each band. Range is **0** to **1000 ms**.

Stereo Analyze

In this algorithm you can look at the two channels of a stereo signal, and also their inversions, sums, and differences. You can adjust their gains, and apply small delays to either or both channels.

L Invert and **R Invert** invert the phase of the channels.

L Out Mode and **R Out Mode** determine which signal is going to be metered and sent to the output of each of the channels. The choices for each are: **L** (left), **R** (right), **(L+R)/2** (normalized sum), **(L-R)/2** (normalized difference), and polarity inverted versions of these.

L/R Delay "time balances" the two signals. At negative values, the right channel is delayed, while at positive values, the left channel is delayed. The range is **-500** to **500 samples**.

RMS Settle controls how fast the RMS meters can rise or fall with changing signal levels. Range is **0** to **300 dB/second**.

The Stereo Image algorithm borrows some features from this algorithm and some from Mono to Stereo, and provides a stereo correlation meter.

FXMod Diagnostic

This algorithm allows you to view the current levels of any data sliders, MIDI controls, switches, or internally generated VAST LFOs, ASRs, FUNs, etc. which are available as modulation sources. It 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 that 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 an FX preset and studio containing the algorithm, then go to one of the FXMod pages of your program or setup (with the studio selected). Select the FXBus that contains the FX preset 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 reenter the studio and Multieffect Editor where you can view the meters on parameter page 2.

Bipole1 through **Bipole4** attach bipolar modulation sources (those that can go positive or negative) to the bipolar meters. The parameters are not adjustable.

Monopole5 through **Monopole8** attach monopolar modulation sources (can go positive only) to the monopolar meters. The parameters are not adjustable.

Special Topics

Song Mode

We've referred to the special aspects of KDFX in Song mode elsewhere in the manual. Here is a summary of what's been said, along with some new information.

FX Mode in Song Mode

If FX Mode on the Effects-mode page is set to **Program**, and you put the K2600 into Song mode, control of KDFX remains with the program that is on the current FX Channel, as set on the Effects-mode page. However, if FX Mode is set to **Auto**, and you put the K2600 into Song Mode, KDFX is under the control of the program that is on the effect channel *within* the song—from Song mode press **Edit** to go to the EditSong:COMMON page, and look at the EffectChan parameter. In fact, when FX Mode is set to **Auto**, the program on the song's EffectChan controls KDFX *any* time a song is playing, even if you're not in Song mode.

The effect channel is saved with the song, so make sure that it is set so that the program that you want to be in control of KDFX is active on that channel.

Clock Setting for tempo-based parameters

The clock setting in Song mode determines how tempo (BPM)-based parameters behave. If you are using the K2600's internal tempo clock, the Clock parameter on the MISC page in Song mode must be set to **Int**. If you are using an external MIDI timing source, such as a sequencer, then that parameter must be set to **Ext**.

If you are using Tap Tempo as a tempo source, then the Clock parameter must be set to **Internal**.

Recording a Setup in Song Mode

In order to record a setup into the internal sequencer, you first need to set the RecTrk parameter on the MAIN Song page to **Mult**. (If you are using an external keyboard that sends on only one MIDI channel, you must also go to the RECEIVE page in MIDI mode and turn on the LocalKbdCh parameter, setting it to agree with the MIDI Out channel of your keyboard.)

There is a complication, however, if you want to record FXMods while you're recording your track using the setup. The sequencer will faithfully record all of your controller movements, but when it plays them back, it plays them as part of a single track. That track contains a program, not a setup, because the sequencer doesn't know how to play setups.

Since all of the FXMods were part of the setup, even though the data on the track is being played back, it's not going anywhere, and no FXMods are being played.

The solution is to have a *program* playing back the data you've just recorded, which has the identical KDFX studio and FXMods as the setup you used to record with. The data has been recorded on the MIDI channel of Zone 1 of the setup. Find the program on that channel, press **Edit** to get inside it, and then import the KDFX studio from the setup you used originally—that's what the ImpFX function is for.

Now save that program (in the same location or a new one) and go back to the song. Make sure that the program's channel and the song's effect channel agree. The FXMods should now all play correctly.

Studios and Songs in Master Mode

When you are using KDFX in Master mode, the studio does not get saved with the song, since it is not a dependent object of the song, or of anything in it. Therefore, if you want to recall a studio whenever you load in a particular song, you either have to:

- Remember to recall the studio by hand, or
- Go into FXMode: Program, and save the studio as part of a program which is on one of the tracks of the song.

Remember, just because you are in FXMode:Program doesn't mean you *have* to use FXMods—you can leave the studio alone, just as you would in Master mode.

KB3 Effects

In the K2600's KB3 mode, KDFX processing is being used to its utmost. Several studios have been included for use with KB3. These studios have a special characteristic: the FXBuses and Aux bus are used together, forming a "virtual" single processing algorithm with 7 (count 'em) PAUs.

There are two algorithms that are designed for this special use: Algorithm **779 KB3 FXBus** (to be used on an insert FXBus), and Algorithm **780 KB3 AuxFX** (to be used on the Aux bus). They don't work very well individually, and should probably not be used that way.