KDFX V2

Musician's Reference

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Chapter 1
Introducing KDFX Version 2

Overview

KDFX V2 gives you more control than ever over the effects in each studio, through several enhancements:

- You can assemble chains of up to four FX Presets on an FXBus.
- You have access to 26 new KDFX algorithms (29 for K2500).
- You can mute or bypass effects and EQ for individual FXBuses. You can now keep effects bypassed outside the program editor, as well.
- You can get to Effects Mode directly from Program, Setup, and Song modes—you don't have to go through another editor first.

If you are familiar with Version 1 of KDFX, you will notice that there are several new pages and parameters giving you access to the new features. You'll also notice some changes to existing pages. The INPUT, AUXFX, and OUTPUT pages have not changed at all with KDFX V2, however.

The following tables give a brief explanation of the major additions and changes in KDFX V2.

Page	Description	What's Different
KDFXMode:MAIN	Displays studio assignment. Also, for each insert FXBus, displays FX Preset assignment, bypass/mute, and PAU assignment.	Was EffectsMode page; controlled studio assignment, FX Mode and FX Chan, and dither (noise-floor adjustment).
EditStudio:FXBUS	New soft buttons for adding/deleting FX Presets on current FXBus; new soft button for bringing up the new FXSEND page.	PAU assignment is handled for each FXBus individually; Level and Balance for each FXBus are part of new FXSend page; New feature: you can chain multiple Presets together on an insert bus using the Add and Remove buttons.
EditStudio:FXSEND	Directs the send of each Insert FXBus to either the default stereo Mix bus or the AuxFX bus, with balance control for each.	Relocates Lvl and Bal parameters from the FXBus page.
KDFXMode:CTRL	Controls FXCtrl (formerly FX Mode) and FX Chan, as well as dither.	Replaces functionality of EffectsMode page, which no longer exists.
KDFXMode:EQBYP	Controls EQ mute/bypass for each Insert FXBus.	New feature.
KDFXMode:FXBYP	Controls independent active/bypassed status for each Insert FXBus and the AuxFX bus.	New feature.
KDFXMode:BUSMUT	Controls independent active/muted status for each Insert FXBus and the AuxFX bus.	New feature.

Additions and Changes to Effects Mode Pages

Prior to V2, there was a single Effects Mode page, which was responsible primarily for setting the FX Mode and FX Chan parameters. This pair of parameters affected how the K2600 / K2500 selected studios when you changed programs, setups, or songs. You would get to the EffectsMode page by pressing the **Effects** button while in a performance mode (not while in an editor). While inside an editor, the Effects button was used to globally bypass or enable the effects.

There's much more going on with V2, so we've added several pages, and moved a number of parameters into more logical groupings. Pressing the **Effects** button now takes you to the KDFXMode:MAIN page, which gives you easy access to the Effects Mode pages.

The Studio Editor has some new soft buttons and pages to accommodate effect chaining in KDFX V2. The **Add** and **Remove** buttons on the FXBUS page let you create and modify chains on the buses. The FX Send and Aux Send fields and graphics that were previously shown on the FXBUS page are now found on their own new page, FXSEND. Pressing the right **more**> button on the FXBUS page reveals the **FXSEND** soft button, as well as the familiar **AUXFX** and **OUTPUT** buttons.

New Button Presses for Bypassing/Enabling Effects

As a result of the new flexibility we've added, bypassing or enabling effects now takes an extra button press or two, because you have to press **Effects**, then **BypAll** or **Enable**. Also, you may need to press **Exit** or **Effects** again, depending on whether you're just toggling the effects on and off, or whether you are leaving the effects bypassed and returning to an editor. See page 1-4 for information on the new bypass features.

MAIN Page

The KDFXMode:MAIN page gives you a summary view of the current effects configuration, including the current studio, the FX Presets assigned to each of the five effects buses, and the bypass status of each bus.

```
KDFXMode:MAIN     FXCtrl:Auto       <>Enable
Studio:113 PltEnvFl4T Plate         Free:0
FX1    43 Plebe Chamber         -   Size:1
FX2    902 Synth Env Filter     B   Size:2
FX3    735 Bap ba-da-dap        -   Size:1
FX4    0 None                   B   Size:0
Aux    103 BigPredelayPlate     B   Size:3
MAIN  CTRL   EQBYP  FXBYP  BUSMUT Enable
```

Figure 1-1 Effects mode: the KDFXMode:MAIN page

As with every other page, the top line of the KDFXMode:MAIN page identifies the page you're on. It also shows you two other important features of Effects mode:

FXCtrl: Effects-control status. Prior to V2, this was called FX Mode. The functionality hasn't changed, but we renamed it because it's now a parameter on the Ctrl page, which is accessible with the **CTRL** soft button.

Enable state: Shows whether KDFX is currently enabled or if any part of KDFX is bypassed or muted.

The second line of the display shows the ID and name of the current studio. When you enter KDFX Mode directly (i.e., not through another one of the K2500 / K2600's editors) you can scroll through the displayed list of studios. This allows you to choose a different studio on the KDFXMode:MAIN page. When FXCTRL is set to Master (see page 1-7), you can also do this, even when you have entered KDFX Mode from within another editor.

If you select the studio then press the **Edit** button, you'll go to EditStudio:FXBUS page, where you can make changes to each bus within the studio.

The second line also shows the number of PAUs available for the current studio ("Free:" on the right-hand side). This number will be 0–4, since in each studio four PAUs are available for the four insert FXBuses (the AuxFX bus has its own fixed set of three PAUs).

The next five lines show the IDs and names of the FX Presets assigned to the five effects buses (insert FXBuses 1–4 and the AuxFX bus). You can't change these assignments on the KDFXMode:MAIN page; to do that you would highlight the Studio name (line two of this page) then press **Edit**. This takes you to the Studio Editor, on the appropriate FXBUS page for the first bus. As in KDFX V1, use the Chan / Bank buttons to move between buses.

Each of these five lines also indicates the bypass status for the five buses, as well as the number of PAUs used by each FXBus. A dash (–) indicates active / enabled, and B indicates bypassed / disabled. You can change the bypass status for a bus by moving the cursor to this field and changing it with either the alpha wheel or pressing one of the increment / decrement buttons.

The size of each FX Preset is measured in PAUs (processor allocation units). FXBuses 1–4 can all use up to four PAUs, but the studio can use a maximum of four total PAUs. The AuxFX bus can use up to three PAUs independent of the insert FXBuses.

Soft Buttons in Effects Mode

The **MAIN** button takes you to the KDFXMode:MAIN page, where you can view the current studio and the FX Presets assigned to the five KDFX buses.

The **CTRL** button takes you to the KDFXMode:CTRL page, which contains parameters that determine which studio gets selected when you select a program, setup, or song. You also use the KDFXMode:CTRL page to change dither and digital word size settings.

Soft Buttons: Configuring Bypasses

You can individually bypass any of the EQ and effects inputs, and also mute any of the FXBuses (the four insert FXBuses and the AuxFX bus). In the enabled state, nothing is muted or bypassed. The K2600 / K2500 always starts up in the enabled state.

Use these soft buttons to perform bypasses and muting:

EQBYP	Displays EQ Bypass page, where you can bypass the EQ on each individual input bus.
FXBYP	Displays FX Bypass page, where you can bypass the effects on individual FXBuses.
BUSMUT	Displays the BusMute page, where you can mute the output of individual FXBuses.
BypAll/Enable	Toggles between enabled state and default bypass state (all buses bypassed, none muted). If you have created a custom bypass scene, **BypAll** resets it to the default bypass state. See page 1-8 for information on creating a custom bypass scene. You may also use either of the **Chan/Bank** buttons to toggle between enabled and bypassed states. This will often be preferable, since **Chan/Bank,** unlike **BypAll**, does not reset the bypass state to the default (all buses bypassed, none muted). Instead, **Chan/Bank** toggles between the enabled state and any custom bypass scene you may have created, allowing you to audition a studio with and without bypasses.

Effects Bus Editor

The revised FXBus Editor lets you create effects Preset chains on any of the four stereo effects buses. See page 1-11 for more information about chaining effects.

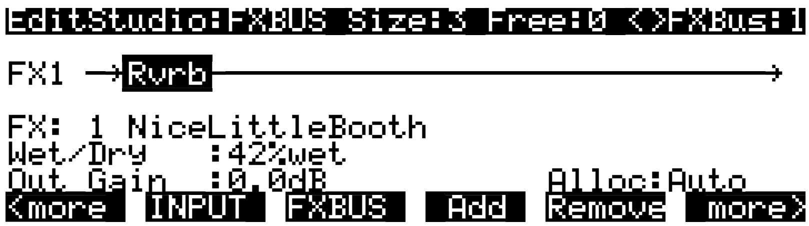


Figure 1-2 Effects Bus Editor display -- single effect

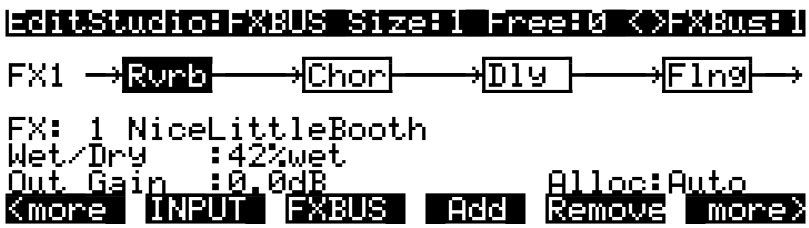


Figure 1-3 Effects Bus Editor display -- four chained effects

The **Add** and **Remove** buttons allow you to define your own chains of effects using up to four FX Presets. The **Add** button creates an effects block (shown as a box) to the right of the current cursor position in the effects chain. You can use a total of four effects in any studio, so if you create a four-block effects chain on a bus then you won’t be able to use any effects on the other buses in that studio. Your K2500 / K2600 keeps track of effects usage for you, and won’t let you add an effects block to a bus if you’re already max’ed out.

The **Remove** button deletes the effects block that the FXBUS editor cursor is on. Adding and deleting effects blocks may cause audio glitches in any signal path and should not be done during critical listening.

Each FX Preset in an effect chain has two “override” parameters (BusMods) that are displayed when that FX block is selected. By selecting the name of an override parameter (e.g., Wet / Dry), you can scroll to choose from any other available parameter.

Each effect also has its full complement of real time modulators as defined and displayed in the Program and Setup editors.

Effects Send Page

The FXSEND page lets you send the output of each stereo effects bus to the stereo mixdown and auxiliary buses.

In KDFX V1, these sends were on the EditStudio:FXBUS page.

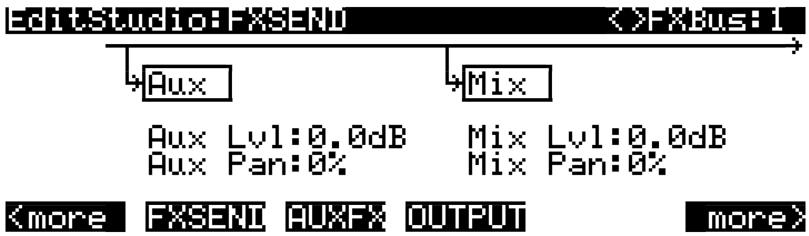


Figure 1-4 Effects Bus Send display

The CTRL Page

The CTRL page is where you set a variety of important parameters for KDFX, including the Effects Control Mode (FXCtrl) for the current studio; this parameter was known as FX Mode in KDFX V1, and its function is the same in KDFX V2.

As in KDFX V1, the CTRL page is also where you set the FX Channel and set the Dither level for the current studio. Depending on the options you have installed, this is also where you'll be able to change word length for digital output.

```
KDFXMode:CTRL     FXCtrl:Auto      <>Enable
Studio:49 Sndboard Room Hall
FXCtrl:Auto
FXChan:Current

Dither:Medium
MAIN   CTRL    EQBYP  FXBYP  BUSMUT Enable
```

Figure 1-5 Effects Control page

Effects Control Mode (FXCtrl)

FXCtrl determines how the K2500 / K2600 selects studios as you change programs or setups, and determines whether you have real-time control over studio parameters—in other words, whether FXMods are active.

If the value of FXCtrl is **Program** or **Auto**, then as you change programs in Program mode, the K2500 / K2600 also loads the studio linked with that program. This activates all the FXMods defined within the program. If the value of FXCtrl is **Setup** or **Auto**, then as you change setups in Setup mode, the K2500 / K2600 also loads the studio linked with that setup. This activates all the FXMods defined within the setup. If the value of FXCtrl is **Auto**, and the value of FX Chan is **Current**, then when you're in Program, Setup, Quick Access, or Song mode, programs, setups and songs automatically use their corresponding studios. In Program, Setup, and Quick Access modes, the studio corresponds to the current program or setup. In Song mode, the studio corresponds to the program on the song's assigned effects channel (which is determined by the value of the EffectChan parameter on the COMMON page in the Song Editor).

If the value of FXCtrl is **Master**, changing programs or setups does not load an associated studio; the current studio is defined by the Studio parameter on the Effects Mode page. Any FXMods defined in the current program or setup are inactive.

Effects Control in Embedded Editors

In the parlance of V.A.S.T., an embedded editor is an editor that you enter while you are already in another editor. An example of this would be entering the KDFX Studio Editor while you are already in the Program Editor. In this sort of situation, an editor may function differently than if you had entered it directly from a performance mode.

When you enter the KDFX Studio Editor from within another editor (for example, you are already in the Program Editor when you press the **Effects** button), KDFX will revert FXCtrl to Auto and FXChan to Current if you attempt to perform an operation that the software doesn't support. For example, you cannot change a program's assigned studio by pressing the **Effects** button to enter the KDFX Editor while you are already within the Program Editor. If you set FXCtrl to Master you will be able to audition different studios, but the software will not let you change a studio. The correct way to change the studio used by a program is to press the **KDFX** soft button from within the Program Editor.

Bypass and Mute pages

In KDFX V1, you could perform a global bypass by pressing the **Effects** button while inside an editor. You could not bypass effects outside of an editor.

With KDFX V2, you can bypass effects buses, inside or outside of an editor, by pressing the **Effects** button followed by either the **BypAll** soft button or one of the **Chan/Bank** buttons. Although similar, the two methods are slightly different:

- The **BypAll** soft button globally bypasses all effects buses, and also resets the default bypass state to bypass all buses. When you press this soft button it changes into the **Enable** soft button, allowing you to toggle between the state where all buses are bypassed and the state where all are enabled.
- Either **Chan/Bank** button toggles between the enabled state and the current bypass state. The current bypass state is either the default (all buses bypassed) or the custom bypass scene you have created. See the section that follows for information about creating a bypass scene.

Pressing the **Effects** button again, or pressing **Exit**, puts you back where you were.

Creating a Custom Bypass Scene

You create a custom bypass “scene” (e.g., effects bypassed on one bus, but not on the other three) by using the soft buttons on the EQBYP, FXBYP, and BUSMUT pages to isolate sounds or effects. You can then toggle between an all-enabled state and your custom scene by pressing either of the **Chan/Bank** buttons (to the left of the display) while in KDFX Mode.

The system indicates whether anything at all is bypassed or muted by showing “Bypass” at the far right of the top line on the display; if nothing is bypassed, this field shows “Enable.” Any settings from the FXBYP page are also indicated on the KDFXMode:MAIN page as either a “B” (bypassed) or a “**-**” (enabled). EQ Bypass and Bus Mute settings, however, are not indicated on the KDFXMode:MAIN page. If you exit this mode with anything bypassed, the **Effects** button's red LED stays lit to remind you that something is not active.

The EQBYP Page

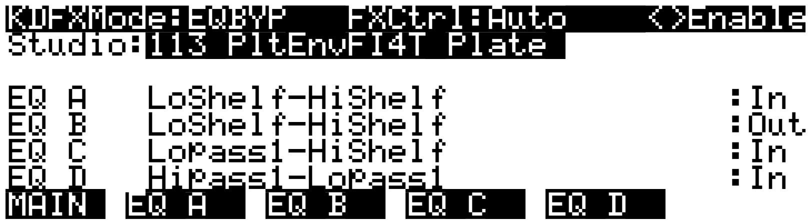


Figure 1-6 EQ Bypass Page

The **MAIN** soft button takes you to the KDFXMode:MAIN page. The soft buttons **EQ A, EQ B, EQ C,** and **EQ D** toggle the bypass / active status for the EQ on the corresponding input buses.

The EQBYP page looks a little different when there are mono inputs to the studio. In this case, press the **L/R** soft button to toggle between left and right mono inputs for a bus.

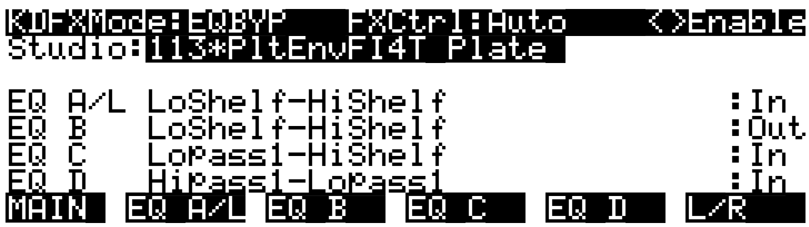


Figure 1-7 EQ Bypass Page with Mono Inputs

The FX Bypass Page

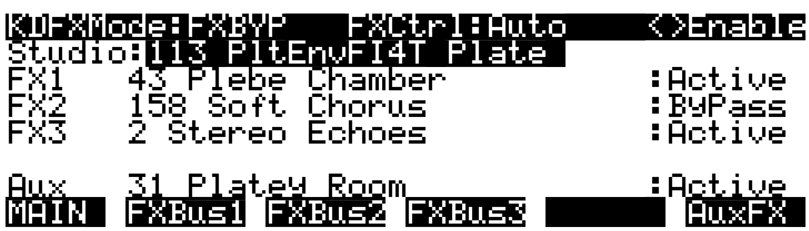


Figure 1-8 FX Bypass Page

The **MAIN** soft button takes you to the MAIN page. The **FX1–FX4** and **AuxFX** soft buttons toggle Bypass / Active status for the effect on the corresponding bus.

The Bus Mute Page

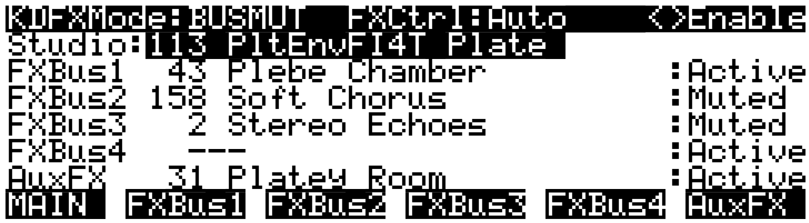


Figure 1-9 Bus Mute Page

The **MAIN** soft button takes you to the KDFXMode:MAIN page. The soft buttons **FXBus1**–**FXBus4** and **AuxFX** toggle the mute / active status for the corresponding input buses.

Chaining Effects

One of the most powerful new features in KDFX V2 is effects chaining, which allows you to send a signal through four consecutive KDFX effects. The screen below shows an example of this:

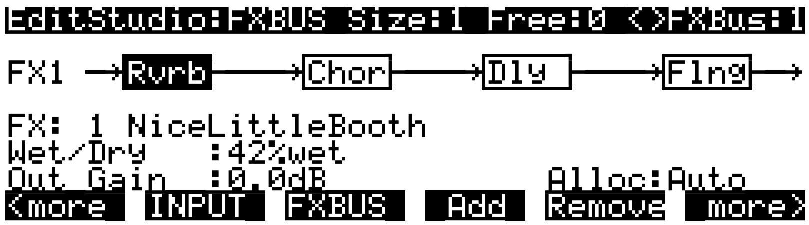
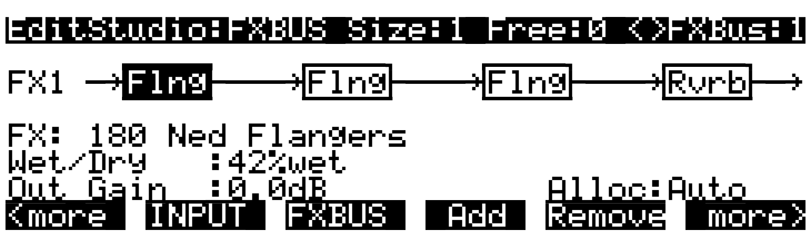


Figure 1-10 Effects Bus Editor display -- four chained effects

Effects chaining allows the 4 PAUs of processing shared among Buses 1-4 of a Studio to be used in series. You can chain one FX Preset into another, into another, up to four in a row, until you run out of PAUs. This is done by removing processing “blocks” from one bus, and adding them to another. As no effect is less than 1 PAU, and only 4 PAUs are available across Buses 1-4, any Studio may have a maximum of 4 blocks, arranged however you please, in which to select Presets (not counting the Aux bus which is unaffected by chaining).

The FXBUS page has changed in KDFX V2 to allow for effect chaining. The chained effects are shown at the top of the display (underneath the top menu line). As an example, start from Program Mode, press the **Effects** button, then select Studio 700 Flanger Trio:

You can see that this studio has three flange effects, followed by a reverb. The effects are numbered 1a through 1d to indicate that they are all part of FXBUS 1, instead of four separate effects buses. Now press **Edit** to go into the Studio Editor. The top of the display shows the four effects chained together. Each block contains an abbreviation based on the algorithm used by the Preset:



The name of the FX Preset for the currently highlighted block is now shown underneath the signal path graphics. In this example, you will see the FX Preset Ned Flangers if the first block is highlighted. You still have 2 Bus Overrides (or Bus Mods) per block, which appear just below the name of the Preset.

Use the left and right cursor buttons to select each block. When a block is selected, move the alpha wheel or press the + or **-** buttons to select a different FX Preset (you can also change the Preset by cursoring to the full name of the Preset after the FX: label).

Notice that the unhighlighted blocks have a box around them. This shows they are active. Since this studio has 4 blocks, each block can use only 1 PAU. If you select an effect that uses more than one PAU, one of the blocks will become inactive and the box surrounding that block will disappear. For example, if you change the first block to FX Preset 183 NarrowResFlange, the box around block 4 disappears. The top line of the display shows you this FX Preset uses 2 PAUs. As in the past, if the Allocation parameter is set to Auto, the lower number blocks have precedence, so block 4 is the one that becomes inactive. If you highlight block 4 at this point, you will see the FX Preset shown in parenthesis, again showing it is not active.

The **Chan/Bank** buttons still move you through the four FXBUSes as in KDFX V1. Since no effects are available in this case, you will see a line with no blocks on them if you look at any bus except FXBUS 1. You can still use a bus to send another signal to the AUX without the chain, by the way, since KDFX has been designed to offer you maximum flexibility.

Gain Staging in Effects Chains

When chaining Presets together, it is sometimes necessary to adjust the levels between blocks, most often to pad the level going into the next block to prevent unwanted clipping. While most algorithms have both an In Gain and an Out Gain parameter, In Gain is not selectable as a Bus Mod. In fact, any Preset beyond the first in a chain cannot use In Gain, and will display the value inside the Preset in parentheses. We suggest, when necessary, choosing Out Gain as a Bus Mod to adjust the output level of an effect, instead of trying to pad the input of the following effect. Of course, you can always edit FX Presets directly and customize them for your chain.

Checking Out Some Chains

For examples of studios with chains, check out studios 700-719. By setting the FX Ctrl parameter (KDFXMode:CTRL page) to Master, and the OutPair parameter (MIDIMode:Channels page) to KDFX-A, you can scroll through Programs on a given MIDI channel and audition these studios as they were intended to be heard, with a variety of input source material.

KDFX V2 Objects

Listed below are the new KDFX Objects added with version 5.*xx* (K2500) or 3.*xx* (K2600) of the Base Objects file. These include new FX Algorithms, Presets, and Studios added to the K2500/K2600 with the new object files. Objects marked “Additional XX new to K2500” were previously available in the K2600 but not in the K2500.

New FX Algorithms

138 Degen Regen BPM	792 Gate+TubeAmp
139 Switch Loops	914 Reverse LaserVrb
140 Moving Delay	915 Gated LaserVerb
161 Allpass Phaser 3	916 Poly Pitcher
741 Rotor 1	917 Frequency Offset
742 VC+Dist+HiLoRot2	918 MutualFreqOffset
743 Subtle Distort	919 WackedPitchLFO
744 Quantize+Alias	920 Chaos!
745 Pitcher+Miniverb	948 Band Compress
746 Reverb+Compress	949 CompressDualTime
781 St Chorus+Delay	971 3 Band EQ
784 St Flange+Delay	972 HF Stimulate 1
790 Gate+Cmp[EQ]+Vrb	975 HarmonicSuppress

Additional FX Algorithms new to K2500

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

New FX Studios

700 Flanger Trio	710 Blues/R&R Guitar
701 Thin Cloud Layer	711 Clean Rhythm Gtr
702 Fazetortion	712 Sweeping Verb
703 Ultraphasulate	713 Pad Ambience
704 Chutney Squishy	714 Coming of Dawn
705 BuddMeetsTomita	715 Wide Raindrops
706 2 Buses 2 Chains	716 Robot Voice
707 RingMod Envelope	717 Before The Crash
708 Drum Megashaper	718 Piano Multi-Verb
709 Dist Lead Guitar	719 Mastering Studio

Additional FX Studios new to K2500

167 auxChorEnvF Hall
168 auxChRvEncr Chor
169 auxChrDstEQ Room
170 aux ChDlSRS Hall
171 aux EQFlng DstEQ
172 auxFlngPhsr Lasr
173 auxFlShQFlg Hall
174 auxFlgDist+ Room
175 aux GtVbFl4T Bth
176 auxRvRvQFlg Hall
177 auxRvRbShapeChmb
178 auxSpinMDly Room
179 aux SweepEchoBth
180 auxRoto&DsFDRPlt
181 auxRot&Ds2FDRPlt
182 auxFlgChDl Hall
183 auxDstLsr CDR
184 CPDlEnFltCmpGtRv
185 RotoOrgFX2 Hall
186 ChDlFlPtLzVb Plt
188 CDR FlgRvb Hall
190 DistRoom GrphEQ
191 Enh Ch 4T Hall
192 FiltCmpExpFl CDR
193 LzVbFlDstEQ Room
194 PhseDist Room
195 ChDlyRvFlRv Hall
196 RmRotr&DstChrPlt
197 Clear Studio

New FX Presets

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

Additional FX Presets new to K2500

11 Viewing Booth
122 Reverse Reverb 2
196 Static Phaser
782 VibrChrDstRotor1
783 VibrChrDstRotor2
784 VibChrDstRotor3
785 FullVbChTubeRotr
786 ChorDlyHall 2
787 Flange Hall 2
788 SpeeChorusDeep
789 Fluid Wash
790 VC+DistRotor
800 Sweet Hall
801 Small Hall
802 Medium Hall
803 Large Hall
804 Big Gym
805 Bright Plate 1
806 Opera House
807 Live Chamber
808 Bathroom
809 Med Large Room
810 Real Room
811 Drum Room
812 Small Dark Room
813 Small Closet
814 Add Ambience
815 Gated Reverb
816 Reverse Reverb
817 Non-Linear
818 Slapverb
819 Full Bass
820 Room + Delay
821 Delay Big Hall
822 Chorus Room
823 Chorus Smallhall
824 Chorus Med Hall
825 Chorus Big Hall
826 Chor-Delay Room
827 Chor-Delay Hall
828 Flange-Dly Room
829 Flange-Dly Hall
830 Stereo Chorus
831 Stereo Flanger
832 Stereo Delay
833 4-Tap Delay
834 Chorus Delay
835 Flange Delay
836 Chorus 4-Tap
837 Flange 4 Tap
838 Chorus Echo
839 Chorus Echoverb
840 Fast Flange
841 Wash
842 Into The Abyss
843 Space Flanger
844 Flange Room
845 Predelay Hall
846 Flange Echo
847 Rotary Club
848 Rotary Hall
849 Chorus
850 Soundbrd/rvb
851 Percussive Room
852 Brt Empty Room
853 Mosque Room
854 New Gated
855 Chorus Slap Room
856 Chorus Bass Room
857 New Chorus Hall
858 Spacious
859 Wash Lead
860 New Hall w/Delay
861 Rich Delay
862 Glass Delay
863 Real Plate
864 Real Niceverb
865 ClassicalChamber
866 Empty Stage
867 Long & Narrow
868 Far Bloom
869 Floyd Hall
870 With A Mic
926 Simple Lazerverb
927 TripFilter
979 Simple Panner
980 Big Bass EQ

Chapter 2
A Tour of KDFX V2

Let's take a tour through several KDFX V2 studios. After you do this, you should have a pretty good idea of what it's like to work with Effects mode.

Note: If you have a K2600, the information in this chapter replaces the information on 9-12 through 9-26 in the Rev. A *K2600 Musician's Guide*; If you're a K2500 user, the information in this chapter replaces pages 2-1 through 2-14 of the *KDFX User's Guide*.

Load the Tutorial files

From the K2600 Demos disk that came with your K2600, load the file **KDFXTUTR.K26** into bank 200...299. K2500 users should load the file **TUTOR1B.K25** from their KDFX Objects disk. You can use a different bank if you like, but then your numbers and the ones in this chapter will be different.

1. Insert the disk into the drive
2. Press **Disk** to go into Disk mode
3. Set Current Disk to **Floppy**
4. Press **Load**
5. Cursor down to **KDFXTUTR.K26** (or **TUTOR1B.K25**)
6. Press **OK**
7. In the "Load this file as:" dialog, select **200...299**
8. Press **OK**
9. Press **OverWrt** (this will erase all objects in the 200s bank, so you should save anything you want to keep before doing this; press **Cancel** if you want to save objects before proceeding). If the 200s bank is empty, then **OverWrt** will not be displayed as an option; in this case, you should select **Append**.
10. Press **Exit** to get back to Program mode.

A Simple Studio

Call up Program #199 on your K2600 / K2500. In Program mode, either scroll the Alpha wheel to Program **199 Default Program**, or press **1-9-9-Enter** on the Alphanumeric pad.

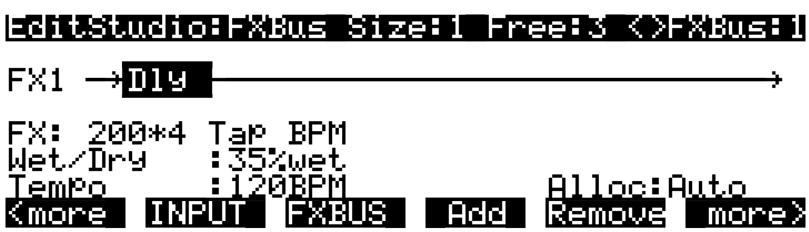
Now go into Effects mode by pressing the **Effects** button, and you'll see this page:

Now press the **CTRL** soft button. A screen such as this one will appear:

Scroll the Alpha wheel until the Studio parameter's value is **200 Simple**.

The FXBus Page

Now press **Edit**, and this page appears:



This is the FXBus page for FXBus1. It is where an FX preset is assigned to the FXBus. Put the cursor on the box containing **200*4 Tap BPM**—this is the current FX preset, a four-tap delay whose speed is expressed in terms of tempo (**B**eats **P**er **M**inute). Use the Alpha wheel to scroll through the many other FX presets that come with KDFX. Like all K2600 / K2500 objects, those that are in RAM (like this one) will have an asterisk in their name, and those that are in ROM will not. There are three more FXBuses, which you can view by pressing the **Chan/Bank** buttons. In this studio they are all empty (No Effect).

Go back to **4 Tap BPM** on FXBus1, and play the piano sound from your keyboard. The arrow next to FX1 flashes, showing that there is audio passing through this FXBus. The arrow keeps flashing as long as the FXBus is processing audio.

Below the FX preset selector is a Wet / Dry control, which determines how much of the signal will pass through the FX preset. Below that is a Tempo control, which sets the timing of the delays. These parameters are called “bus overrides,” because they override parameters which are actually inside the FX preset itself—these parameters can be adjusted from inside the FX preset, or they can be set from out here, where they are much more convenient. If you change FX presets, these values change, because their values inside the various FX presets are all different. There will be more about bus overrides a little later in this chapter.

Allocation

The Allocation parameter determines how many processor allocation units (PAUs) are reserved for this FXBus. The number of PAUs an FX preset uses is dependent on the algorithm at the core of the FX preset. Algorithms can use anywhere from 1 to 4 PAUs, depending on their complexity. As you scroll through the FX presets, the number of PAUs required by each one is shown on the top line of the display (Size:), along with the number of PAUs that are available (Free:) for other algorithms.

The 4 insert FXBuses have 4 PAUs to share among them, so if any bus uses more than one PAU, it means that some buses cannot be assigned an FX preset. This is a very common situation, as you shall see. The Aux bus has its own set of 3 PAUs, which are completely independent and are not shared with the insert FXBus PAUs.

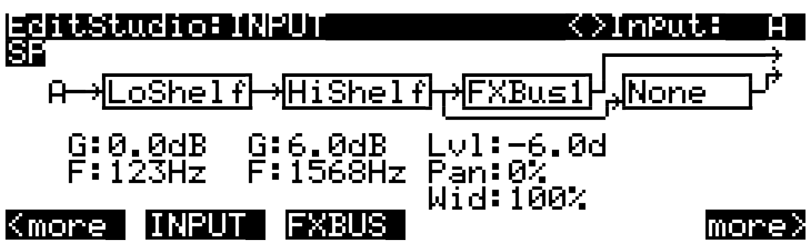
You can preassign a PAU value to an FXBus, in which case any FX presets that require more PAUs than you have given the bus cannot be loaded into the bus. If you try to put an FX preset into a bus that requires more PAUs than are currently available on that bus, the preset’s name appears in parentheses—exactly the way a KB3 program appears if you try to select it on a channel that’s not the KB3 channel.

In most factory studios, the Allocation parameters are set to Auto, in which case PAUs are assigned dynamically as you assign FX presets to the various buses. As you unassign FX presets from buses, or assign FX presets with smaller PAU requirements, the PAUs freed up are automatically reassigned to other buses where they are needed.

A value of Auto for this parameter allocates PAUs to lower-numbered FXBuses first, as needed.

The INPUT Page

Press the soft button labeled **INPUT**. This page appears:



This is the INPUT page. It is showing Input A, as indicated both on the left side and in the upper right corner—use the **Chan/Bank** buttons to view the other three inputs. Input A is the first stereo signal pair coming from the K2600 / K2500 Program Editor’s Output section. Depending on how the K2600 / K2500 is set up, this could be a single layer of a single program, or multiple layers, or multiple programs, or one or more zones from a setup, or the output from one or more MIDI channels.

The **S** at the upper left says that the Input A is being handled as a stereo feed; this can be changed to two mono feeds. The **P** means that the stereo feed has a Pan control; you can choose to make this a Balance control instead, by setting this parameter to **SB**.

If you play on the keyboard, you can see the arrow next to the letter **A** flashing, as audio is being passed through this part of the studio. The arrow on this page flashes only as long as there is an input signal present.

Equalization

The first two blocks are the low and high EQs on the input. Put the cursor on either box and turn the Alpha wheel, and you will see the options you have available for types of EQ—these include **None**, which bypasses that EQ. The first block has more choices than the second.

The G underneath each block is its Gain; **0.0dB** is unity gain; the signal passes through without change. (There is no Gain parameter when a block is set to LoPass or HiPass.) F is the equalizer's frequency. In the input section for this example, the high frequencies are boosted 6.0 dB above 1568 Hz.

Sends

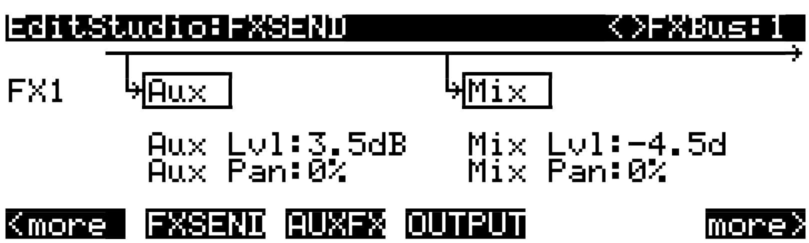
The third and fourth blocks determine the destinations of the Input A signal: each block can be set to route the signal to any of the four insert FXBuses, or to None. You cannot, however, set both blocks to the same destination.

The Lvl control is the FXBus send; it sets the level of the signal to the FXBus above it. In this example, the level is backed off 6.0 dB, to compensate for the treble boost in the equalizer, so that the signal doesn't overload the FXBus.

Pan determines the position of the signal respective to the left and right sides. Width (which is not shown when you are using mono inputs) determines how much the left and right sides' signals will be separated or blended.

The FXSEND Page

Press the **more>** soft button to display additional soft buttons for this page. Then press **FXSEND** to see the FXSEND page:

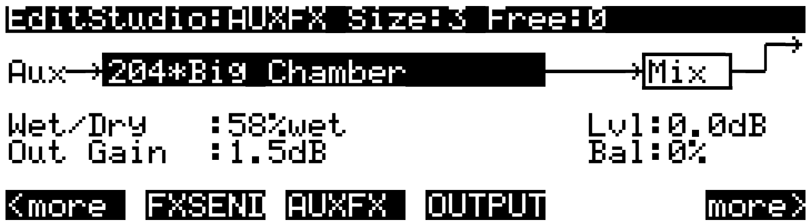


Levels

There are two sets of level and pan controls from the FX preset to the output mixer. Use the first set, Aux Lvl and Aux Pan, to specify how much of the sound will go to the global or Auxiliary effects bus, and how its two channels are panned. On this FXBus, the signal to the Aux bus is boosted 3.5 dB. The second set determines how much of the sound goes to the Mix bus. On this FXBus, the Mix signal is attenuated -4.5 dB.

The AUXFX Page

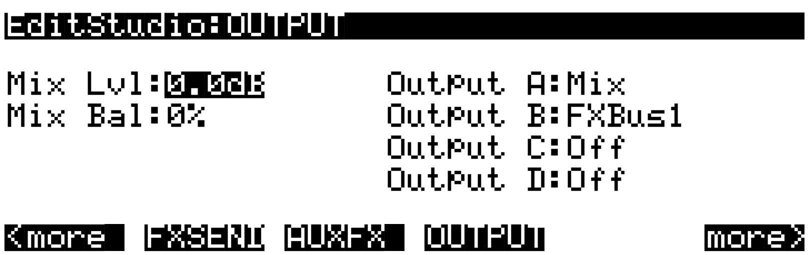
Press the **AUXFX** soft button. This page appears:



This is the Global AUXFX page, and shows us what is happening on the Auxiliary effects bus. The Aux bus is a second processor, which follows the four insert FXBuses. It has its own FX preset, with bus overrides, and level and balance controls to feed it into the Mix bus. It doesn't share PAUs with the FXBuses; it has three PAUs of its own, and consequently doesn't have an Allocation parameter. The Aux bus can be routed all by itself to an output, as we'll see. In this studio, the Aux bus contains a chamber reverb.

The OUTPUT Page

Press the **OUTPUT** soft button.



The OUTPUT page is the interface to the real world. It determines which of the signals going through the various effects buses show up at the K2600/K2500's four sets of *physical* outputs: A, B, C, and D. These four outputs, all stereo, are both analog and digital (through the KDS bus). Output A also goes to the K2600/K2500's AES/EBU digital output.

In this studio, Output A is carrying the Mix, that is, the combination of the outputs of the four FXBuses (only one of which is in use) and the Aux bus. Output B is carrying FXBus1, which is the signal after it passes through the delay on FXBus1, but before it gets to the reverb on the Aux bus. The other outputs are carrying no signal.

***Note**: If you're using the Mix audio outputs, keep in mind they carry the summed signals of audio outputs A through D. Normally you would assign each of the audio outputs differently. In the example above, you might set Output B to* **Off**, *or Output A to* **AuxFx**, *to avoid applying the FXBus1 effect to the entire program.*

Here's a diagram of what this studio looks like:

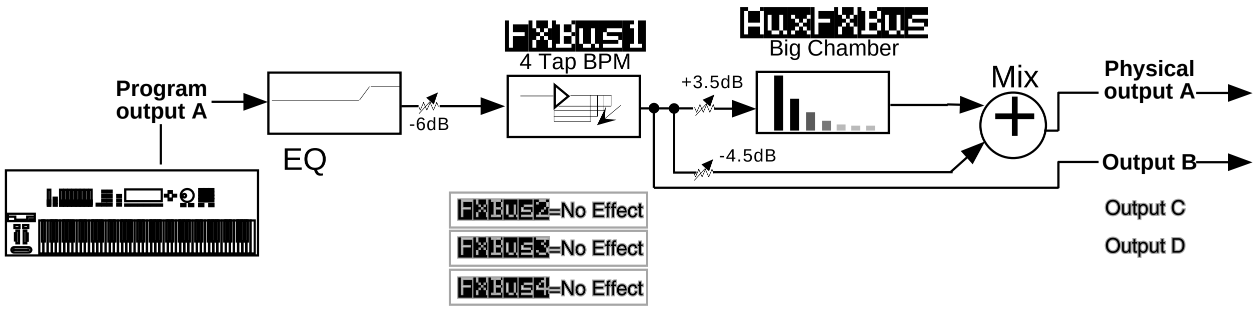


Figure 2-1 Structure of Studio 200*

A More Complex Studio

Press **Exit** as many times as necessary to get to Program mode. Call up Program **200 ElecPno/Flute**. This is a split keyboard program: On Layer 1, which has been assigned to the KDFX-A outputs, is an electric piano, whose key range goes up to B4. On Layer 2, which goes to the KDFX-B outputs, is a flute, whose key range starts at C5.

Press the **Effects** mode button, then the **CTRL** soft button and set FXCtrl to **Master**. Then call up Studio **201 RngMd/PFD/Plt**. Press **Edit** to look inside of this studio.

FXBus1

On the first FXBus is an FX preset called **201*Tut Ring Mod**. The algorithm this uses is a ring modulator, which is a processor that takes the sounds coming into it and combines them with static waveforms by adding and subtracting their frequencies, thereby creating interesting nonharmonic effects. Notice that this FX preset uses 1 PAU.

```
EditStudio:FXBUS Size:1 Free:1 <>FXBus:1

FX1   →RMod──────────────────────────────→

FX1: 201*Tut Ring Mod
Wet/Dry    :90%wet
Out Gain   :0.0dB              Alloc:Auto
<more  INPUT  FXBUS   Add   Remove more>
```

As you play on the lower part of the keyboard, the arrow next to FX1 flashes, but as you play on the upper part it doesn't. That's because the upper part of the keyboard (the flute sound) is routed to a different FXBus.

Bus Overrides

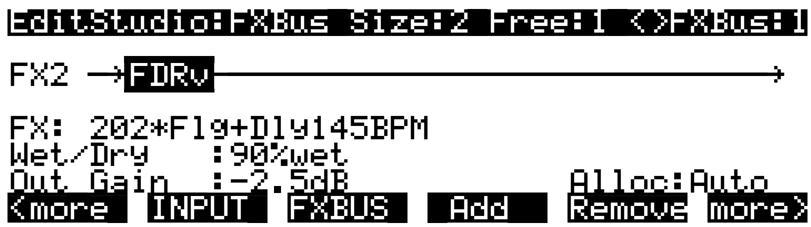
There are two FX preset parameters on this page: Wet / Dry mix and Output Gain. These parameters actually exist inside the FX preset, and are placed on this page so you can control them without editing the FX preset itself. These are bus overrides, as described earlier. You can change both the value of the override parameter *and* the name of the parameter that shows up in the bus override: to select a different parameter, simply highlight its name and scroll the Alpha wheel. As you do so, you will see the other parameters inside the FX preset that can be brought out to this page.

These overrides (that is, which parameters are available, and their values) are stored as part of the studio, not as part of the FX preset, and therefore you don't have to create new FX presets just because you want to change a couple of parameters. There are two bus overrides available for each of the four insert FXBuses and the Aux bus.

If you don't want any parameter control on this page, select the parameter's names, then scroll the Alpha wheel until you see **None**.

FXBus2

Press the **Chan/Bank Up** button to get to FXBus2:



On this bus is an FX preset called **Flg+Dly145BPM**, which uses a combination algorithm that has flanging, delay, and a reverb all rolled into one. The **145BPM** part refers to the fact that the delay times are based on a tempo of 145 BPM,

As you play the flute sound, the arrow next to FX2 flashes, and it keeps on flashing as long as the various feedback delays are sounding. It doesn't flash when you play on the lower part of the keyboard.

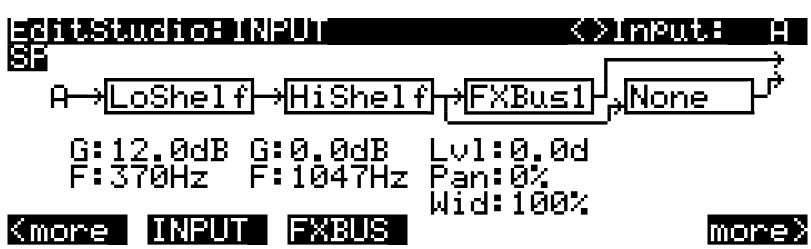
This FX preset uses 2 PAUs. Along with the 1 PAU in use on FXBus 1, this makes 3 of the 4 available PAUs accounted for, so the Free parameter has a value of **1**.

Press **more>**, then **FXSEND** to see that the output configuration of this FXBus has the signal going to the Aux bus attenuated by -9.5 dB, and going to the main Mix bus at unity gain.

The other two FXBuses are empty, which you can confirm by pressing the **Chan/Bank** buttons a few times.

Inputs

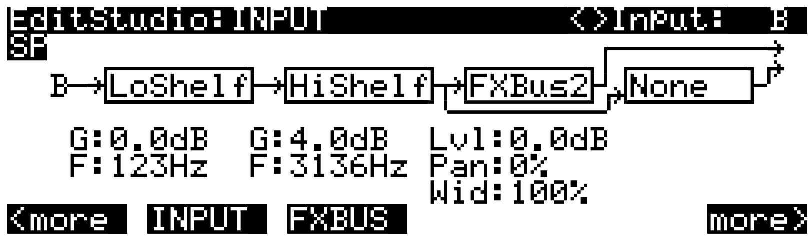
Now let's look at the inputs to the FXBuses. Press **<more**, followed by the **INPUT** soft button, and see this page:



Input A carries the electric piano, coming from the program's KDFX-A outputs. Play on the piano part of the keyboard, and the arrow next to A flashes.

This input is configured to be stereo. It has a large bass boost: 12.0 dB of everything at 370 Hz and below, which adds a strong low-frequency emphasis to the signal being ring-modulated. Its signal is being sent only to the first FXBus. The stereo separation (Width) of the signal is at maximum.

Use the **Chan/Bank Up** button to go to Input B.

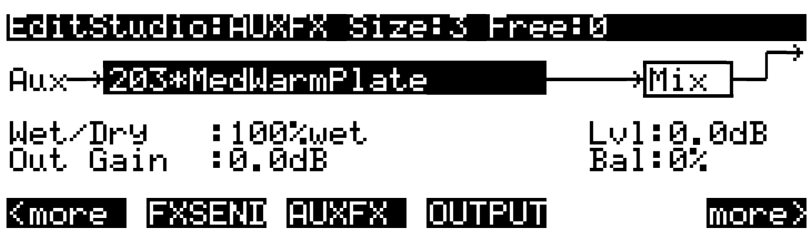


This is the flute, coming from the program's KDFX-B outputs. It is also stereo. The incoming signal has a strong treble boost on it. It is sent directly to FXBus2 at unity gain and full width. Play on the flute part of the keyboard, and the arrow next to B flashes.

The other two inputs, C and D, are not assigned to any FXBus.

AuxFX Bus

Now let's look at the Auxiliary FX Bus (Aux bus). Press **more>**, then the **AUXFX** soft button to look at its page.

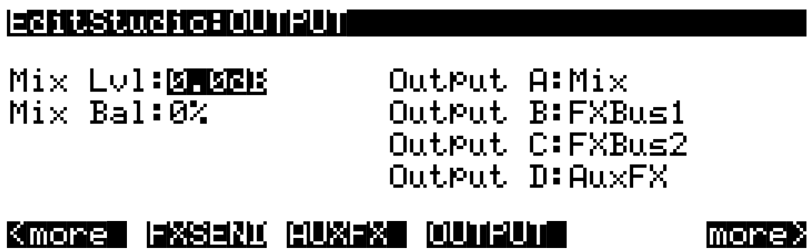


Here is an FX preset called **MedWarmPlate**, which is just what it sounds like: a medium-sized, warm-sounding plate reverb. It has two bus overrides, Wet / Dry mix, and Output Gain. The Wet / Dry level is set to 100%, because the reverb can be applied to any of this studio's FXBuses. You set the actual wet / dry mix for these with the Aux and Mix levels on the FXSEND pages. By keeping the reverb level on the Aux bus set to 100%, you avoid mixing in a non-reverbed signal twice.

Since both Insert FXBuses have signal going to the Aux Bus, the arrow next to Aux will flash as long as any signal processing is going on in either of the insert FXBuses.

Outputs

Finally, press the **OUTPUT** soft button to get to the OUTPUT page. Here we see that the four physical output pairs are each passing different parts of the studio. If the outputs are connected to an external mixer, you can treat each of them separately: recording them on different tracks of a tape deck, sending them to different outboard processors, or mixing them differently in a monitor mix.



Output A has the Mix bus. This is the combined output of the two FXBuses, plus the reverb on the Aux bus. Its gain and balance are at unity.

Output B has the output of FXBus1, that is, the ring-modulated piano, without any reverb.

Output C has the output of FXBus2, the delayed / flanged / flute, without any reverb.

Output D has the output of the Aux bus, which is *just* the reverb signal, with no dry component, since the value of Wet / Dry on the AUXFX page is **100%**.

Note*: If you're using the Mix audio outputs, keep in mind they carry the summed signals of audio outputs A through D. Normally you would assign each of the audio outputs differently. In the example above, you might set Outputs B through D to* ***Off****, or Output A to* ***Off****, to avoid overlapping assignments.*

Here is the overall structure for this studio:

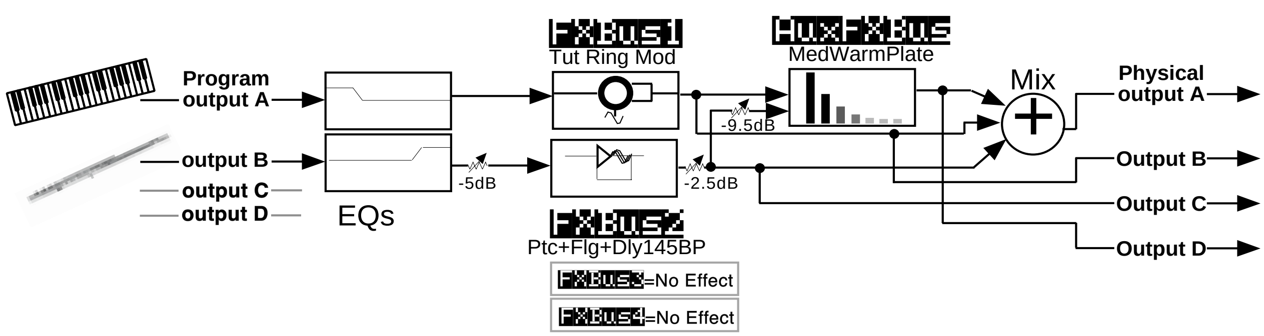


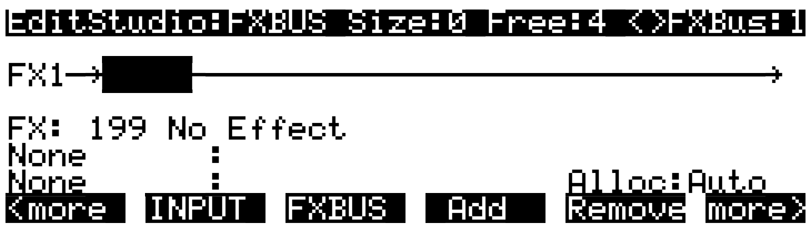
Figure 2-2 Structure of Studio 201

Chaining Effects

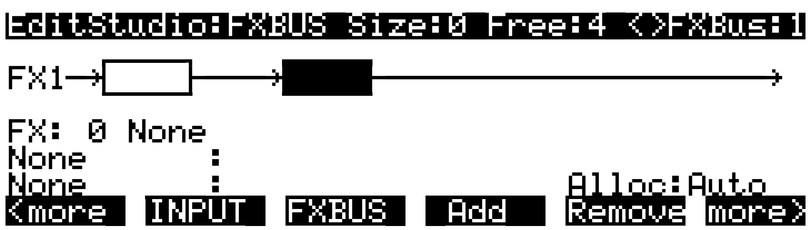
Now we'll see how you can employ multiple effects on a single bus using the new chaining feature in KDFX V2.

Start in Program Mode by highlighting Program **199 Default Program**, then pressing the **Effects** button. From the KDFXMode:MAIN page, select Studio **199 Default Studio** (highlight the Studio name, then scroll through the list or press **1-9-9-Enter** on the Alphanumeric pad).

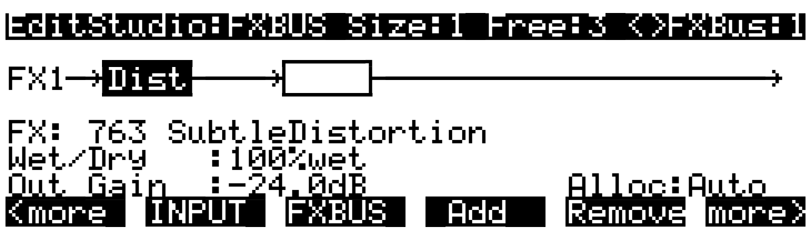
With the Studio name still highlighted, press the **Edit** button to enter the Studio Editor. As shown below, you can see that there is an empty effects block on this bus and all 4 PAUs are still available. If you'd like, you can press the **Chan/Bank** buttons to see that while there are other empty blocks (although not on FXBus 4, since it is generally used as a dry bus or a pass-through to the AUX FX), there are no effects on any of the buses in this studio.



Now press the **Add** soft button to add another empty effects block to this bus:



We'll use a distortion preset for the first effect in this block. Use the cursor keys to highlight the first effects block, then select Preset **763 SubtleDistortion**:



You can see that the effects block now displays an abbreviation that indicates the type of effect, while the lines below show the name of the specific effect preset, as well as the bus overrides defined for this preset. As always, you can select one of the bus overrides and change it to be any parameter of the effect's underlying algorithm. You'll probably want to keep the two overrides here, however, since they offer useful control over the preset: Wet/Dry lets you quickly adjust how much of the signal is effected, while Out Gain lets you boost or cut the audio level of this effect as it enters the next effect in the chain.

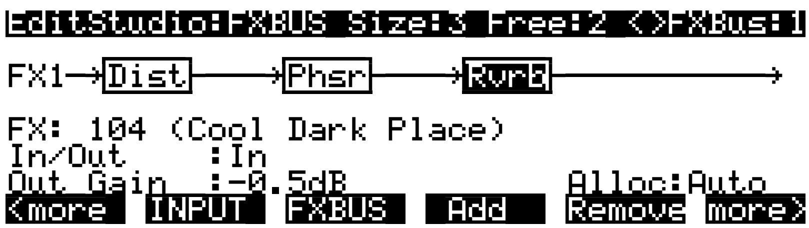
We'll now add phaser and reverb effects to the chain:

Use the cursor keys to highlight the second effects block, then select Preset **191 Slow Deep Phaser**:

Press **Add** again to add another effects block. If the "Out of effect blocks" message displays (and it should), then you'll have to remove an effects block from one of the other buses in this studio. Press the **Chan/Bank** up button to display the screen for FXBus 2; you will see an empty effects block, highlighted for your convenience; press the **Remove** soft button and that block (that you weren't using anyway) is now available for you on FXBus 1.

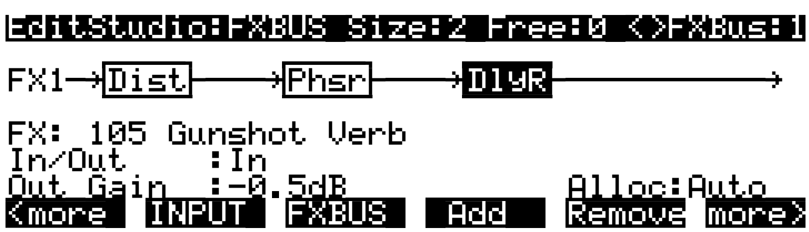
Press the **Chan/Bank** down button to return to FXBus 1. Use the cursor keys to highlight the **Phsr** block, then press the **Add** soft button. This time an empty effects block will be added to the chain for you. Note that the new block is added to the right of the current block, which is why we highlighted the **Phsr** block before pressing **Add**.

While the third block in the chain (the one you just added) is highlighted, select Preset **104 Cool Dark Place**. Your display will look something like this:



The outlined box around the effect block and the parentheses around the preset name indicate that we've chosen an effect that requires more processing power than we currently have available in this studio. Take a quick glance at the top line of the screen and the problem becomes obvious: Cool Dark Place wants 3 PAUs (as shown by the **Size** parameter) but only 2 are available (as shown by the **Free** parameter). KDFX will still let you place the preset on the chain, but it won't be active, so you won't hear the effect on your signal.

Let's see if we can find a reverb that fits in this chain. Press the **+** cursor button once to have a look at the next listed preset, **105 Gunshot Verb**:



As you can see, this effect requires 2 PAUs, which is what happens to be available in the Studio. If you decide to use this effect, however, you will not be able to add any more presets to this chain, since you have now maximized your PAU usage. If you'd like to have four effects on the chain, you'll have to select only effects that use a single PAU. (You'll also have to remove the empty effects block from FXBus 3 so that it will be available for FXBus 1.) Keep in mind, however, that you can still send the output of this chain to an effect on the Aux bus, which allows effects using up to 3 PAUs.

Now let's audition this effects chain: play your keyboard to hear the sound of the **199 Default Program** running through the 3 effects you've chosen in series.

To use this effects chain with any other program, you'll have to save the studio. Press **Exit**, then **Rename.** Enter a name for the studio, then press **OK**. The save dialog now offers you the next available studio number; you can save the studio at this location or choose a different one. Press **Save** and you are returned to the KDFXMode:MAIN page. Press **Exit** to return to ProgramMode. You can now select a program, press **Edit**, followed by the **KDFX** soft button (you'll have to press one of the **more** soft buttons a few times to bring the KDFX soft button into view), and select your new studio for the program's effects. Don't forget to check the program's OUTPUT page to be sure that its output pair is routed to KDFX-A.

A Complex Studio with Real-Time Control

The final studio we'll look at is a bit more complex, not least because it's under real-time control. Getting real-time control of a studio requires doing some advance work. Most noticeably, you have to create a program or setup that uses a particular studio, then define a set of FXMods in that program or setup. The FXMods provide the settings that link physical controllers such as sliders to studio effects like wet / dry mix.

For all of this to work, the FXCtrl parameter on the KDFXMode:CTRL page must be set to a value of **Program** (if you want only programs to be in charge of studio selection and controls) or **Setup** (if you want only setups to have this ability)—or **Auto** if you want to control over studios whenever you select a program *or* a setup.

In this example, we'll work with *setup* control over a studio—the procedure for working with program control is almost exactly the same.

Setting the FXCTRL

To enable real-time control of the studio from the setup, we have to put the K2600 / K2500 in the correct Effects mode. Go to the KDFXMode:CTRL page (press **Effects** followed by the **CTRL** soft button), and set FXCtrl to **Setup**. FX Channel automatically goes to **None**.

The Setup

Now let's look at the setup that we're going to use to control our studio. Press **Setup** to go into Setup mode, and select **200 KDFXCombo**.

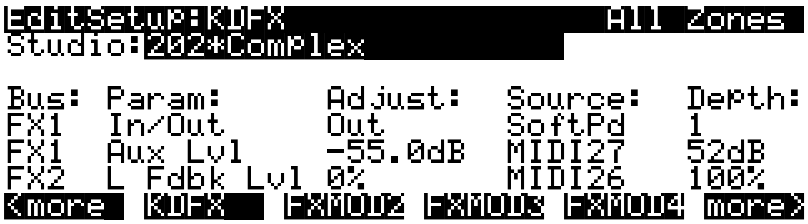
This is a four-zone setup, with bass and drums at the bottom, electric piano in the middle, and a breathy flute-like sound on the top. Each layer goes to a separate KDFX output pair, so they can all get different processing.

If you edit the setup and look at the various zones, you'll see this:

- Zone 1: **Gtr Jazz Band**, a layered bass and drum program, going to KDFX-A.
- Zone 2: **Dual Slap Bass**, also going to KDFX-A. This and the previous zone are active from the bottom of the keyboard up to A3.
- Zone 3: **Pno & Epno & Pad**, an electric piano and pad program, going to KDFX-B, which is active from A$^{\#}$3 to F$^{\#}$5.
- Zone 4: **Hybrid Vox**, going to KDFX-C, and active from G5 and up.

Looking at the Studio

Starting from SetupMode, press **Edit**, then **more>** three times. Now press the **KDFX** soft button to view the studio. The name of the studio associated with this setup, **202*Complex**, appears.

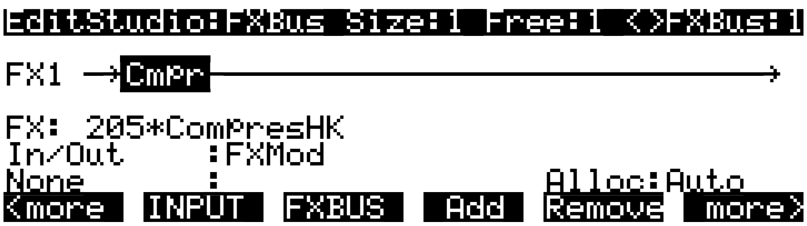


Highlight the studio's name and press **Edit**, and let's dig into this studio.

Press **INPUT** if you'd like to look at the EditStudio:INPUT pages. These are all set up straightforwardly, with Input A going to FXBus1, Input B going to FXBus2, etc. Press the **Chan/Bank** buttons to move between the input groups.

On the FXBus Pages

Press **FXBUS** to look at the FXBuses. On FXBus1 is **205*CompresHK**, a hard-knee compressor.



The In / Out parameter shown on this page is a bus override, similar to the ones we've seen in earlier tutorials. However, instead of saying **In** or **Out** it says **FXMod**. This means that this parameter isn't controlled from inside the studio at all—it's controlled by something outside the studio.

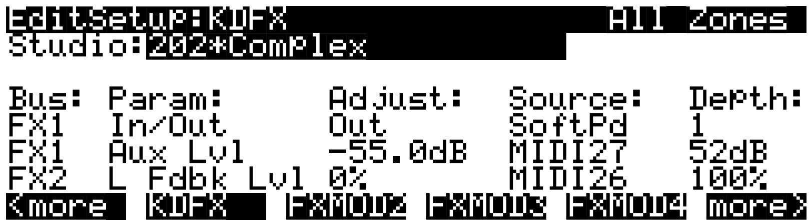
As it happens, it's controlled by the soft pedal, MIDI Controller 67—Switch Pedal 3 for K2600 / K2500 keyboard users. We'll see how this is done in a moment. Pressing this pedal causes the compressor to kick in, squashing the dynamic range of the sound. The MakeUpGain inside the compressor is set to 6.0 dB, however, so the level doesn't change much when the compressor is engaged.

Such control is useful on this program, which uses velocity-switching to change drum sounds. As you play harder, the drum sounds change, but they also get louder, which you may not want. With the compressor engaged, the drum sounds can change *without* getting louder.

There's another FXMod, which is assigned to the Aux bus send level; this is indicated on the EditStudio:FXSEND page (press **more>** followed by the **FXSEND** soft button to have a look). On the Aux bus is a reverb, so this FXMod controls how much of the signal coming through here will go to the reverb. It's under the control of Slider G, MIDI Controller 27. Play the bass and drums and move that slider, and you'll hear the reverb go in and out.

Setting Up FXMods

How do you set up FXMods? That’s back on the KDFX page in the Setup Editor. Exit the Studio Editor (if you have made any changes, don’t save them—that will only confuse things!), then press the **KDFX** soft button. Listed here are the FXMods for this setup:



The first FXMod affects the signal on the FX1 bus, as shown by the Bus parameter. Param indicates which parameter is affected on the FX1 bus—in this case, the In / Out parameter. The Adjust parameter defines the initial condition of the parameter controlled on this bus—in this case, the In / Out parameter has an initial value of **Out** (disengaged).

The Source parameter determines what MIDI Controller affects the In / Out parameter. For the first FXMod it’s **SoftPd** (that’s MIDI 67, which is Soft Pedal according to the MIDI specification, and which the K2600 / K2500 uses as the default destination for Switch Pedal 3). While this setup is current, *any* K2600 / K2500 controller—be it a wheel, slider, ribbon, or button—that’s programmed to send MIDI 67 will affect the In / Out parameter on the FX1 bus of this studio.

The Depth parameter defines how much the soft pedal affects the In / Out value, and here it is **1**.

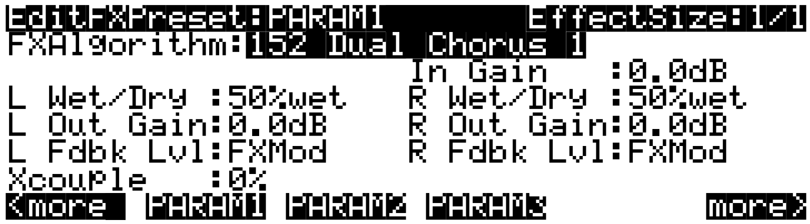
For parameters with binary value, likes **In** / **Out** or **On** / **Off**, the Depth parameter can have only three values: **1**, **0**, or **-1**. If it is set to **1** or **-1**, then changing the state of the Source (in this case, depressing the soft pedal), changes the parameter’s state—in this case, going from Out to In. (If it’s set to **0**, then the Source has no effect on the value of the parameter.)

Now look at the second FXMod. It sets the Aux Lvl on the FX1 bus to an initial value **-55 dB**. The Source, **MIDI 27**, can raise that level by as much as 52 dB, to put it at -3 dB. You can hear this in action when you move Slider G (which defaults to MIDI 27) while you play this patch.

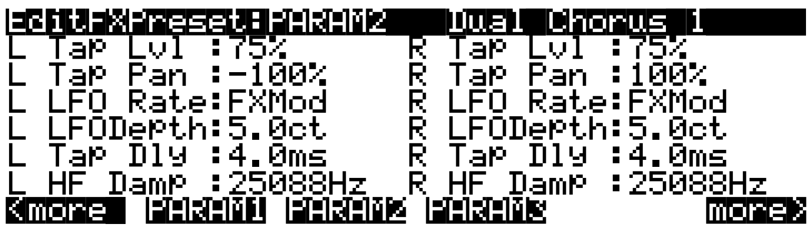
FXBus 2

Let’s go back into the Studio Editor and look at FXBus 2. Highlight the studio name on the EditSetup:KDFX page, then press **Edit**; then press the upper **Chan/Bank** button to change buses. Here’s our piano and string pad, going through an FX preset called **206*Fast&RichChorus**.

If you now go inside the FX Preset (by pressing **Edit**), you'll see that the left and right feedback levels ("Fdbk Lvl") are under FXMod control:

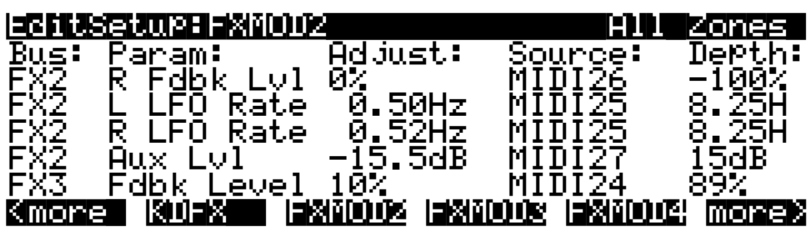


Press **PARAM2**, you'll see that the Left and Right LFO Rates are *also* under FXMod control.



Go back out of the FXPreset and the Studio, to the Setup's **KDFX** page, by pressing **Exit** twice. Here we see that on FX Bus 2, L Fdbk Lvl has a starting value of 0%, and can be changed, using MIDI controller 26 (Slider F), up to 100%.

Press **FXMOD2** to go to the next page of FXMods, and you'll see the same slider changing the R Fdbk Lvl, only in this case the Depth is -100%, meaning the feedback on this channel will be out of phase with the main signal.

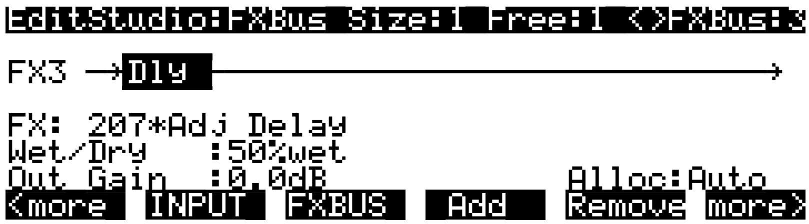


Further down the FXMOD2 page are the assignments to the LFO rates: MIDI controller 25 (Slider E) is assigned to the left and right LFO1 Rates, with minimum values of 0.50 and 0.52 Hz, respectively, and maximum change of 8.25 Hz. Set the feedback level high and you can really clearly hear the LFO rate changing (but watch out that the effect doesn't go into oscillation at the highest feedback level).

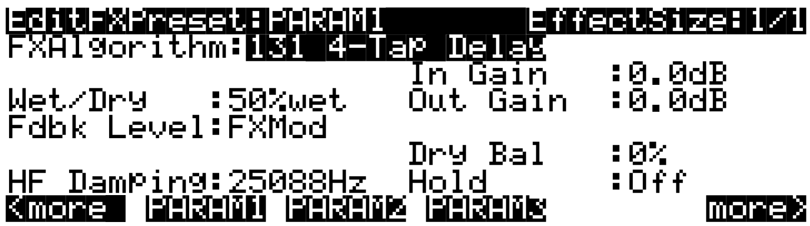
Finally, MIDI Controller 27, our old friend Slider G, controls the Aux Level send on this bus as well, controlling the amount of this signal that will be sent to the Aux reverb, and thus how much reverb will be applied to the piano / pad sounds. As you can see, you can assign a single Source to multiple FXMods.

FXBus 3

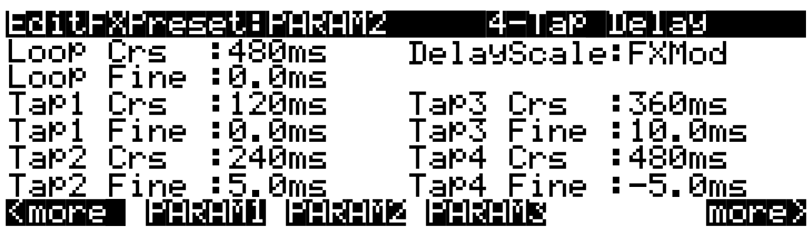
Press **Edit**, then use the **Chan/Bank** buttons to select FXBus 3. Here's where our breathy flute sound is, and it's going through a delay FXPreset called **207*Adj Delay**.



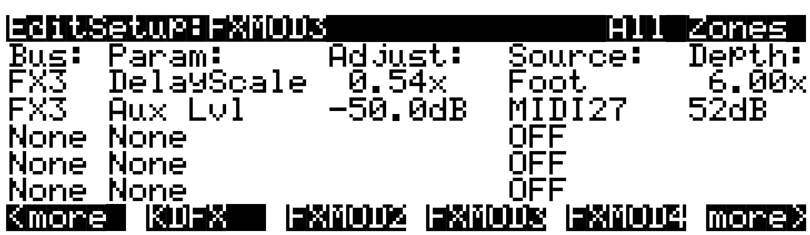
Press **Edit** to inside the FXPreset, and you'll see that the Feedback Level is under FXMod control.



Press PARAM2, and on that page, you'll see that Delay Scale is also under FXMod control:



Go back out to the Setup (press **Exit** twice), press the **FXMOD3** soft button, and look at the FXMOD3 page.



On the FXMOD3 page, we see that the Delay Scale, which scales all of the various delays in the Algorithm, is controlled by the Foot Pedal, MIDI controller 4—Control Pedal 1 for K2600/K2500 keyboard users. The delay is nominally a 480 ms loop, with four equal-spaced taps inside it, 120 ms apart. With the Foot Pedal, we can scale all those times by a factor of between 0.54 and 6.00.

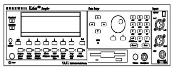
Finally, the Aux Level from this bus, controlling another reverb send, is once again assigned to MIDI controller 27, Slider G.

And in Conclusion...

Here's a summary of the FXMods in our complex studio with real-time control:

- Soft pedal puts compression on the bass and drums.
- Slider F controls the feedback on the chorus on the piano/string pad. Slider E controls the LFO speed on that chorus.
- Slider D controls the feedback on the delay on the breathy flute.
- Foot Pedal controls the delay time on the breathy flute.
- Slider G controls the reverb send for all three zones.

A Note About Effects in Setups



This applies primarily to K2600R / K2500R users who are playing their units with a conventional MIDI keyboard. In order to play setups, which normally require a multi-zoned / multichannel keyboard like the K2600 or PC2, the K2600R / K2500R includes a parameter called Local Keyboard Channel (LocalKbdCh), found on the MIDI Mode RECEIVE page. If you set this parameter to the transmitting channel of the keyboard, then the K2600 takes all incoming MIDI data on that channel and treats it as if it was coming from a local keyboard—that is, it plays all the zones in the setup.

However, when you use this feature, the K2600 will respond to certain MIDI Controller messages coming in on this channel only if the Controller has been specified as a destination on the controller assignment pages in the setup (SLIDER, SLID / 2, etc.)—and then, only if the setup assigns the customary physical controller to that MIDI Controller.

For example, Slider A is customarily assigned to Data (MIDI Controller 6), but if on the setup's SLIDER page, Slider A has been reassigned to something else, then Data messages coming from the external keyboard on the local keyboard channel will not be recognized. Therefore, any MIDI Controller message that is not in its customary place in the setup cannot be used as a KDFX Source, because the setup will not recognize it. A list of the physical controllers affected by using the local keyboard channel is included in the MIDI Mode chapter of your K2600 / K2500 Musician's Guide.

This is not an issue in Program mode, or when using the K2600 / K2500 with a sequencer. In those cases the LocalKbdCh should be set to **None**, and there is no restriction on the MIDI data passing through.

Building Your Own

Now that we've walked you through a complex studio, here are some hints on how to go about building your own.

First, you want to decide whether you want the studio to be static, or to be dynamic under FXMod control from the K2600 / K2500 keyboard and / or an external MIDI source. If it's to be static, the value of FXCtrl must be **Master**, while if it's to be dynamic, the value of FXCtrl must be **Program**, **Setup**, or **Auto**, depending on how you want to approach it.

Then you need to look at how your program outputs are arranged, so you can design the studio intelligently. If you're using internal setups, look at the output assignments on the various zones, and change them so sounds that need different effects are separated, and those that can use the same effects are grouped. If you're using a MIDI sequencer, you might want to use the Channel output overrides in MIDI Mode, and arrange your sequence so that the assignment of tracks to FXBuses is determined by the tracks' MIDI channel assignments.

Now set up your studio, assigning FX presets to the program outputs / KDFX inputs. Many of the ROM studios follow a common organizational plan, which might be a good starting point for your studios:

- FXBus 1 contains a relatively simple reverb with a low Size requirement.
- FXBus 2 contains an effect that does not increase the "length" of the sound (that is, no reverb or delay), something like chorus, flange, phaser, distortion, shaper, pitcher, enhancer, EQ, or EQ morpher.
- FXBus 3 contains effects that take up lots of time, such as delays, delays with reverb, or other "Lead" sounds.
- FXBus 4 is dry, since the first three FXBuses have probably used up all the PAUs.
- The AUXFX 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 FXBus 1 to 0dB or higher), it frees up FXBus1 for use as an Enhancer, Stereo Image, Flanger, etc.

Finally, set up your FXMods in the setup or program you plan to use. If you're using a sequencer, you might want to dedicate a program *just* to the studio and FXMods, with no sound coming from that program.

Keep in mind that FXMods don't always have to be dynamic—they can be used to assign static (Source: OFF) values to the parameters in a studio that are different from the studio's normal parameters. This lets you create and store multiple variations on a studio without making each one a separate studio—when you want to call up the variation, merely call up the program that contains the correct FXMods.

One last reminder: don't forget to save your studio!

Chapter 3
Algorithm Reference

This chapter provides complete reference information for the new algorithms included in KDFX V2. This information is in addition to the algorithm information provided in Chapter 10 of your *K2600 Musician's Reference* or your *K2500 KDFX Algorithm Reference*.

138 Degen Regen BPM

Long delay allowing loop instability

PAUs: 4 each

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

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

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

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

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

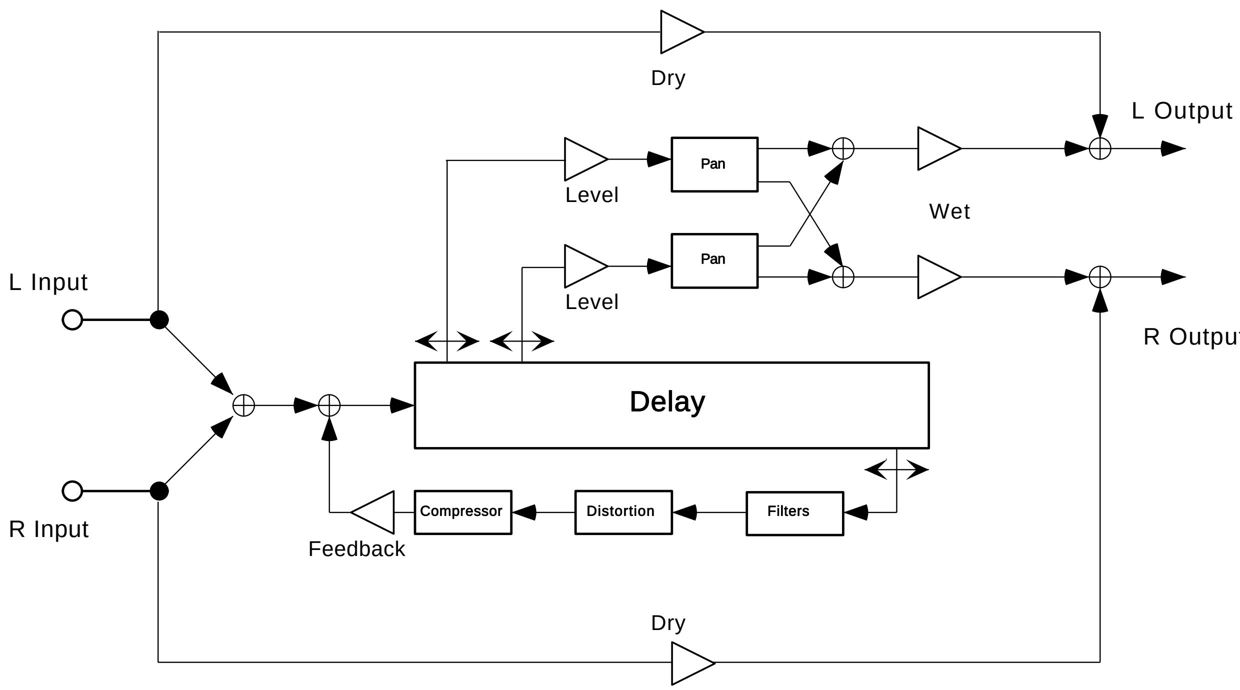


Figure 1 Degen Regen BPM

Parameters:

Page 1

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

Page 2

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

Page 3

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

Page 4

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

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

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

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

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

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

Tempo In **Degen Regen BPM**, Tempo is the basis for the delay lengths, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to **System**, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. In this case, FXMods (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

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

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

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

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

Bass Gain The amount of boost or cut in dB that the bass shelving filter should apply to the low frequency signal components. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency. Since the filters are in the delay feedback loop, the cut or boost is cumulative on each pass the sound makes through the loop.

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

Treb Gain The amount of boost or cut in dB that the treble shelving filter should apply to the high frequency signal components. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency. Since the filters are in the delay feedback loop, the cut or boost is cumulative on each pass the sound makes through the loop.

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

Midn Gain The amount of boost or cut in dB that the parametric filter should apply to the specified signal band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency. Since the filters are in the delay feedback loop, the cut or boost is cumulative on each pass the sound makes through the loop.

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

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

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

LpLFOPhase Specifies the phase angle of the feedback (loop) LFO relative to the output tap LFOs and the system (or MIDI) tempo clock, if turned on (see Tempo). For example, if one LFO is set to **0°** and another is set to **180°**, then when one LFO delay tap is at its shortest, the other will be at its longest. If the system (or MIDI) tempo clock is turned on , the LFOs are synchronized to the clock with absolute phase.

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

T*n*LFOPhase Specifies the phase angle of the output LFO tap (1 or 2) relative to the other output LFO tap, the feedback (loop) LFO tap, and the system (or MIDI) tempo clock, if turned on (see Tempo). For example, if one LFO is set to **0°** and another is set to **180°**, then when one LFO delay tap is at its shortest, the other will be at its longest. If the system (or MIDI) tempo clock is turned on, the LFOs are synchronized to the clock with absolute phase.

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

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

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

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

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

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

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

Dist Drive Applies a boost to the feedback signal to overdrive the distortion algorithm. When overdriven, the distortion algorithm will soft-clip the signal. Since distortion drive will make your signal very loud, you may have to reduce the feedback amount or turn on the compressor as the drive is increased.

DistWarmth A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal.

139 Switch Loops

Looped delay lines with input switching

PAUs: 2

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

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

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

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

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

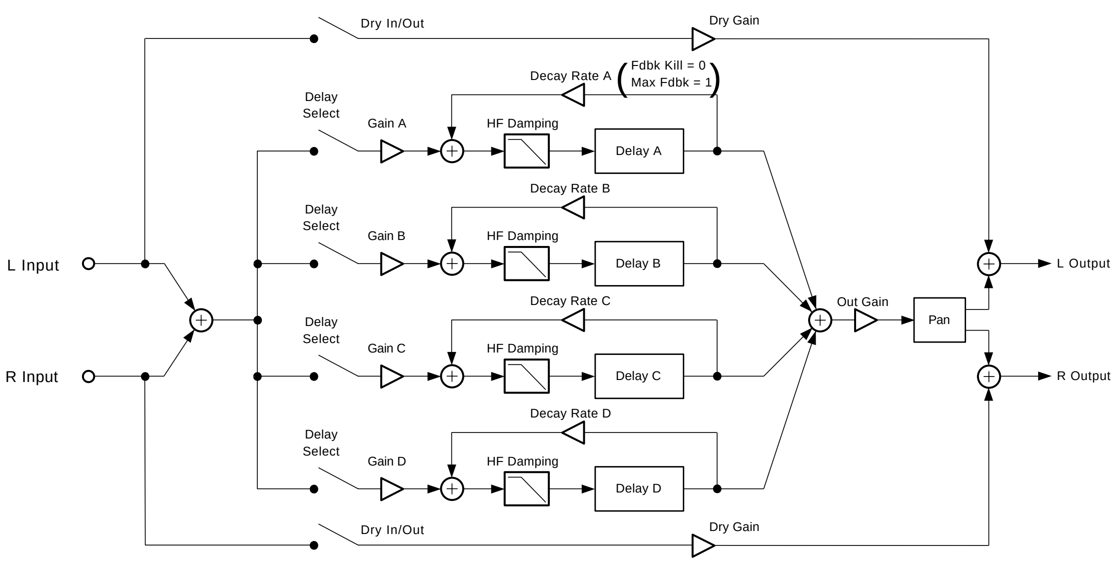


Figure 2 **Switch Loops**

Parameters:

Page 1

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

Page 2

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

Page 3

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

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

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

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

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

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

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

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

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

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

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

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

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

140 Moving Delay

Generic stereo moving delay lines

PAUs: 1

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

Parameters:

Page 1

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

Page 2

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

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

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

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

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

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

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

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

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

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

161 Allpass Phaser 3

Allpass filter phasers

PAUs: 3

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

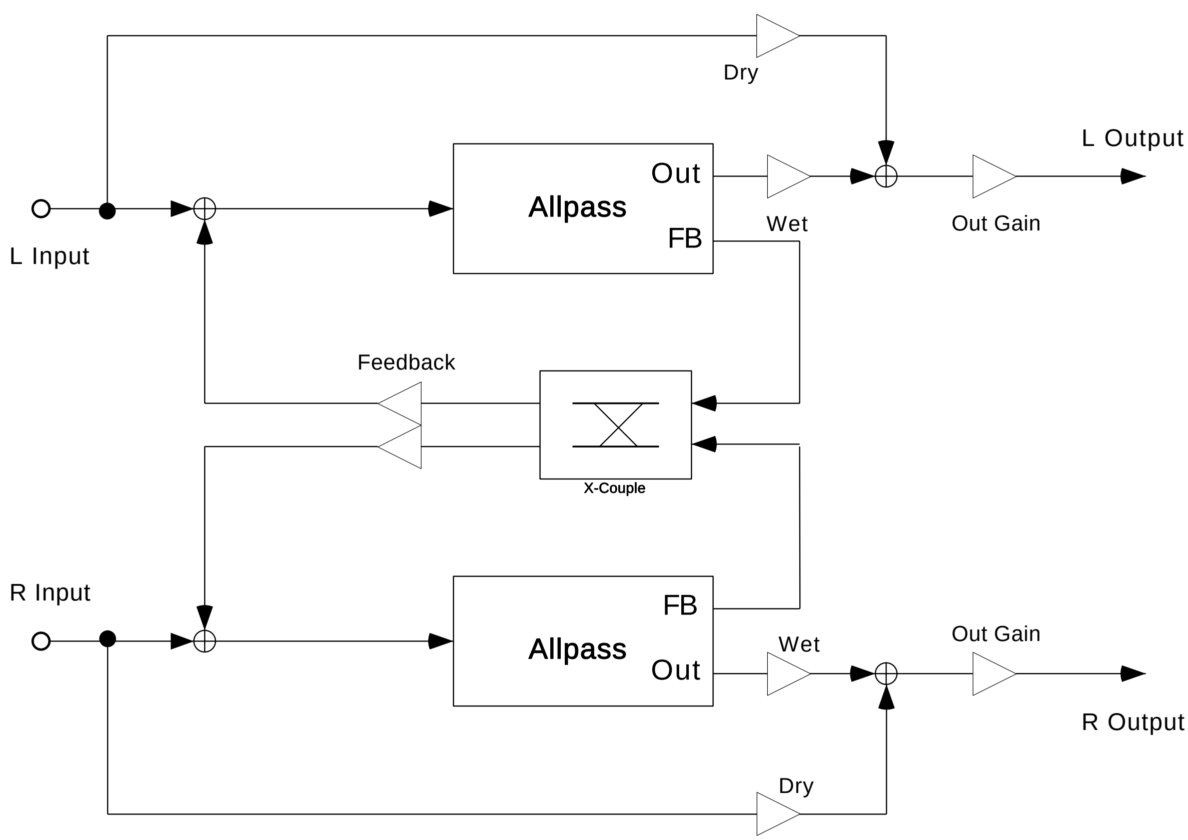


Figure 3 An allpass phaser

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

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

As mentioned earlier, allpass phasers leave the phaser motion up to you, so they have no built in LFOs. To get phaser motion, you have to change the filter center frequencies (left and right channels) yourself. The best way to do this is with an FXMod.

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

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

Parameters:

Page 1

Wet/Dry	-100 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Level	-100 to 100%	XCouple	0 to 100%
LCenterFreq	8 to 25088 Hz	RCenterFreq	8 to 25088 Hz
FB APNotch	3 to 6 or 7 to 10	OutAPNotch	3 to 6 or 7 to 10

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

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

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

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

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

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

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

Rotary Effects

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

Rotating speaker algorithms

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

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

The first effect in the chain is often the Hammond vibrato / chorus algorithm. The vibrato / chorus has six settings which are the same as those used in the Hammond B3: three vibrato (V1, V2, V3) and three chorus (C1, C2, C3) settings. In **VC+Tube+Rotor 4**, the vibrato chorus has been carefully modeled after the electromechanical vibrato / chorus in the B3. The vibrato / chorus in the other smaller algorithms use a conventional design, which has been set to match the B3 sound as closely as possible, but does not quite have the same character as the fully modeled vibrato / chorus.

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

The rotating speaker has separately controllable tweeter and woofer drivers. The signal is split into high and low frequency bands and the two bands are run through separate rotors. The upper and lower rotors each have a pair of virtual microphones that can be positioned at varying positions (angles) around the rotors. An angle of 0° is loosely defined as the front of the speaker. You can also control the levels and left-right panning of each virtual microphone. The signal is then passed through a final lowpass filter to simulate the band-limiting effect of the speaker cabinet.

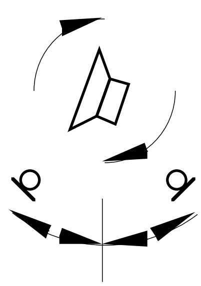


Figure 4 Rotating speaker with virtual microphones

For the rotating speakers, you can control the crossover frequency of the high and low frequency bands (the frequency where the high and low frequencies get separated). The rotating speakers for the high and low frequencies have their own controls. For both, the rotation speed, the effective driver size, and tremolo can be set. The rotation rate sets how fast the rotating speaker is spinning. The effective driver size is the radius of the path followed by the speaker relative to its center of rotation. This parameter is used to calculate the resulting Doppler shift of the moving speaker. Doppler shift is the pitch shift that occurs when a sound source moves toward or away from you the listener. In a rotating speaker, the Doppler shift will sound like vibrato. As well as Doppler shift, there will be some acoustic shadowing as the speaker is alternately pointed away from you and toward you. The shadowing is simulated with a tremolo over which you can control the tremolo depth and “width.” The high frequency driver (rotating horn) will have a narrower acoustic beam width (dispersion) than the low frequency driver, and the widths of both may be adjusted. Note that it can take up to one full speaker rotation before you hear changes to tremolo when parameter values are changed. Negative microphone angles take a longer time to respond to tremolo changes than positive microphone angles.

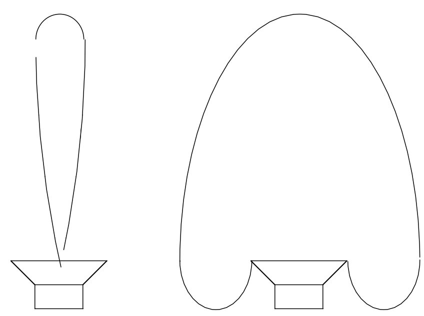


Figure 5 Acoustic beams for (i) low frequency driver and (ii) high frequency driver

You can control resonant modes within the rotating speaker cabinet with the Lo and Hi Resonate parameters. For a realistic rotating speaker, the resonance level and delay excursion should be set quite low. High levels will give wild pitch shifting.

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

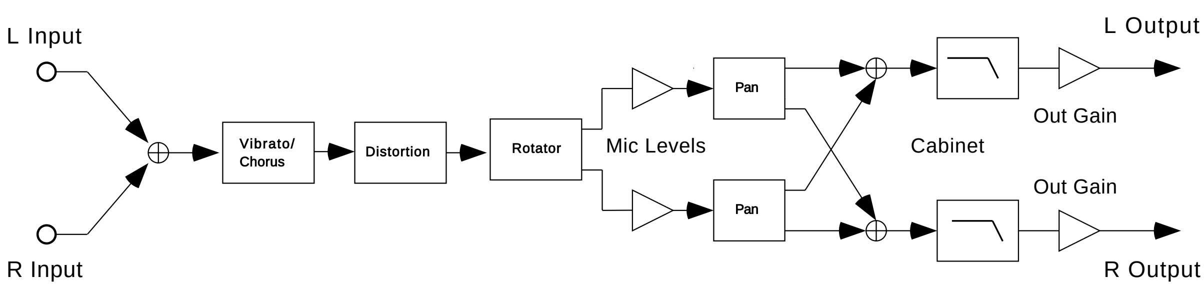


Figure 6 VC+Dist+1Rotor 2

Parameters (VC+Dist+1Rotor 2):

Page 1

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

Page 2

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

Page 3

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

Page 4

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

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

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

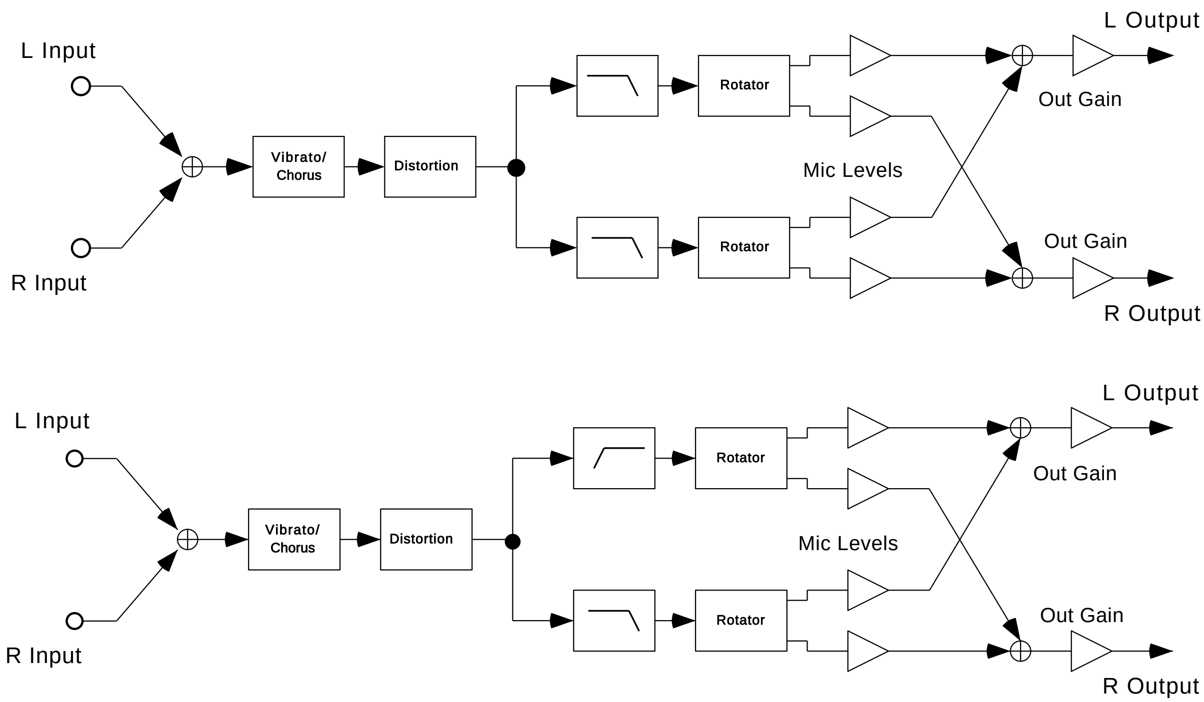


Figure 7 VC+Dist+HiLoRotr and VC+Dist+HiLoRot2

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

Page 1 (VC+Dist+HiLoRotr)

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

Page 1 (VC+Dist+HiLoRot2)

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

Page 2

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

Page 3

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

Page 4

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

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

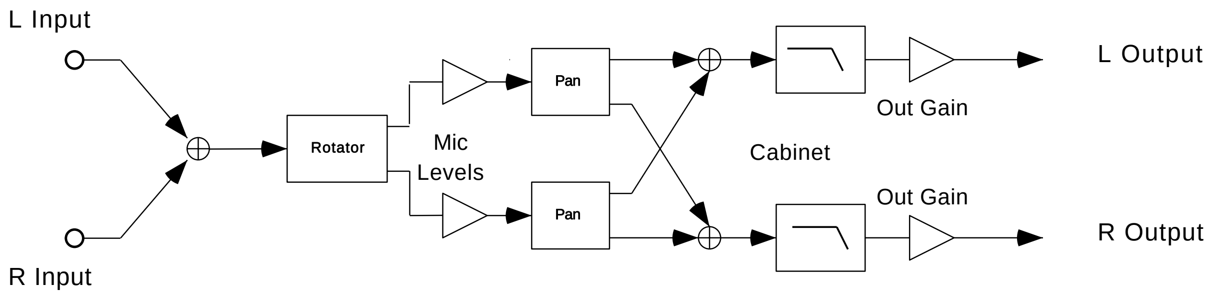


Figure 8 Rotor 1

Parameters (Rotor 1):

Page 1

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

Page 2

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

Page 3

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

Page 4

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

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

Parameters (VC+Tube+Rotor 4):

Page 1 (VC+Tube+Rotor 4)

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

Page 2

Xover	8 to 25088 Hz		
Lo Gain	Off, -79.0 to 24.0 dB	Hi Gain	Off, -79.0 to 24.0 dB
Lo Rate	-10 to 10 Hz	Hi Rate	-10 to 10 Hz
Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%
		Hi Beam W	45.0 to 360.0 deg

Page 3

LoMicA Pos	-180.0 to 180.0 deg	LoMicB Pos	-180.0 to 180.0 deg
LoMicA Lvl	0 to 100%	LoMicB Lvl	0 to 100%
LoMicA Pan	-100 to 100%	LoMicB Pan	-100 to 100%
HiMicA Pos	-180.0 to 180.0 deg	HiMicB Pos	-180.0 to 180.0 deg
HiMicA Lvl	0 to 100%	HiMicB Lvl	0 to 100%
HiMicA Pan	-100 to 100%	HiMicB Pan	-100 to 100%

Page 4

LoResonate	0 to 100%	HiResonate	0 to 100%
Lo Res Dly	10 to 2550 samp	Hi Res Dly	10 to 2550 samp
LoResXcurs	0 to 510 samp	HiResXcurs	0 to 510 samp
ResH/LPhase	0.0 to 360.0 deg		

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

In/Out Gain The overall gain or amplitude at the input or output of the effect. For distortion, it is often necessary to turn the output gain down as the distortion drive is turned up.

VibChInOut When set to **In** the vibrato/chorus is active; when set to **Out** the vibrato/chorus is bypassed.

Vib/Chor This control sets the Hammond B3 vibrato/chorus. There are six settings for this effect: three vibratos **V1**, **V2**, and **V3**, and three choruses **C1**, **C2**, and **C3**.

Roto InOut When set to **In** the rotary speaker is active; when set to **Out** the rotary speaker is bypassed.

Dist Drive or Tube Drive Applies a boost to the input signal to overdrive the distortion algorithm. When overdriven, the distortion algorithm will soft-clip the signal. Since distortion drive will make your signal very loud, you may have to reduce the Out Gain as the drive is increased.

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

DistWarmth A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal.

DistLPFreq Controls one-pole lowpass filters in the **PolyDistort+EQ** (in **VC+Dist+HiLoRotr**). Without the lowpass filters, the sound tends to be too bright and raspy. With less distortion drive, less filtering is needed. If you turn off the distortion curve (set to **0%**), you should turn off the lowpass filter by setting it to the highest frequency.

Cabinet LP A lowpass filter to simulate the band-limiting of a speaker cabinet. The filter controls the upper frequency limit of the output.

Xover The frequency at which high and low frequency bands are split and sent to separate rotating drivers.

Gain The gain or amplitude of the signal passing through the rotating speaker.

Rate The speed of the speaker rotation.

Size The effective size (radius of rotation) of the rotating speaker in millimeters. Affects the amount of Doppler shift or vibrato of signal.

Tremolo Controls the depth of tremolo of the signal. Expressed as a percentage of full scale tremolo.

Beam Width The rotating speaker effect models the acoustic radiation pattern of a speaker ranging from omnidirectional (radiates in directions in equal amounts) to a wide beam. You may adjust the beam width from **45°** to **360°**. If you imagine looking down on the rotating speaker, the beam angle is the angle between the -6 dB levels of the beam. At **360°**, the speaker is omnidirectional.

Resonate A simulation of cabinet resonant modes expressed as a percentage. For realism, you should use very low settings.

Res Dly The number of samples of delay in the resonator circuit in addition to the rotation excursion delay.

Res Xcurs The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker.

Res Phs This parameter sets the relative phases the resonators. The angle value in degrees is somewhat arbitrary and you can expect the effect of this parameter to be rather subtle.

Lo Gain The gain or amplitude of the signal passing through the rotating woofer (low frequency) driver.

Lo Rate The speed of the woofer rotation.

Lo Size The effective size (radius of rotation) of the rotating woofer in millimeters. Affects the amount of Doppler shift or vibrato of the low frequency signal.

Lo Trem Controls the depth of tremolo of the low frequency signal. Expressed as a percentage of full scale tremolo.

Hi Gain The gain or amplitude of the signal passing through the rotating tweeter (high frequency) driver.

Hi Rate The speed of the tweeter rotation.

Hi Size The effective size (radius of rotation) of the rotating tweeter in millimeters. Affects the amount of Doppler shift or vibrato of the high frequency signal.

Hi Trem Controls the depth of tremolo of the high frequency signal. Expressed as a percentage of full scale tremolo.

Hi Beam W The rotating speaker effect models a rotating horn for the high frequency driver. The acoustic radiation pattern of a horn tends to be a narrow beam. You may adjust the beam width from **45°** to **360°**. If you imagine looking down on the rotating speaker, the beam angle is the angle between the -6 dB levels of the beam. At **360°**, the horn is omnidirectional (radiates in all directions equally).

Mic Pos The angle of the virtual microphones in degrees from the “front” of the rotating speaker. This parameter is not well suited to modulation because adjustments to it result in large sample skips (audible as clicks when signal is passing through the effect). There are two pairs of microphones (A and B) for high and low frequency drivers.

Mic Lvl The level of the virtual microphone signal being sent to the output. There are two pairs of microphones (A and B) for high and low frequency drivers.

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

LoResonate A simulation of cabinet resonant modes expressed as a percentage. For realism, you should use very low settings. This is for the low frequency signal path.

Lo Res Dly The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the low frequency signal path.

LoResXcurs The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the low frequency signal path.

HiResonate A simulation of cabinet resonant modes expressed as a percentage. For realism, you should use very low settings. This is for the high frequency signal path.

Hi Res Dly The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the high frequency signal path.

HiResXcurs The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the high frequency signal path.

ResH/LPhs This parameter sets the relative phases of the high and low resonators. The angle value in degrees is somewhat arbitrary and you can expect the effect of this parameter to be rather subtle.

743 Subtle Distort

Adds small amount of distortion to signal.

PAUs: 1

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

Parameters:

Page 1

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

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

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

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

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

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

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

Dist LP A Frequency of Lowpass Filter A.

Dist LP B Frequency of Lowpass Filter B.

744 Quantize+Alias

Digital quantization followed by simulated aliasing.

PAUs: 1

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

Quantization distortion is a digital phenomenon caused by having only a limited number of bits with which to represent signal amplitudes (finite precision). You are probably aware that a bit is a number which can have only one of two values: 0 or 1. When we construct a data or signal word out of more than one bit, each additional bit will double the number of possible values. For example a two bit number can have one of four different values: 00, 01, 10 or 11. A three bit number can take one of eight different values, a four bit number can take one of sixteen values, etc. An 18-bit digital-to-analog converter (DAC) like the one in the K2600 can interpret 262,144 different amplitude levels (2^{18}).

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

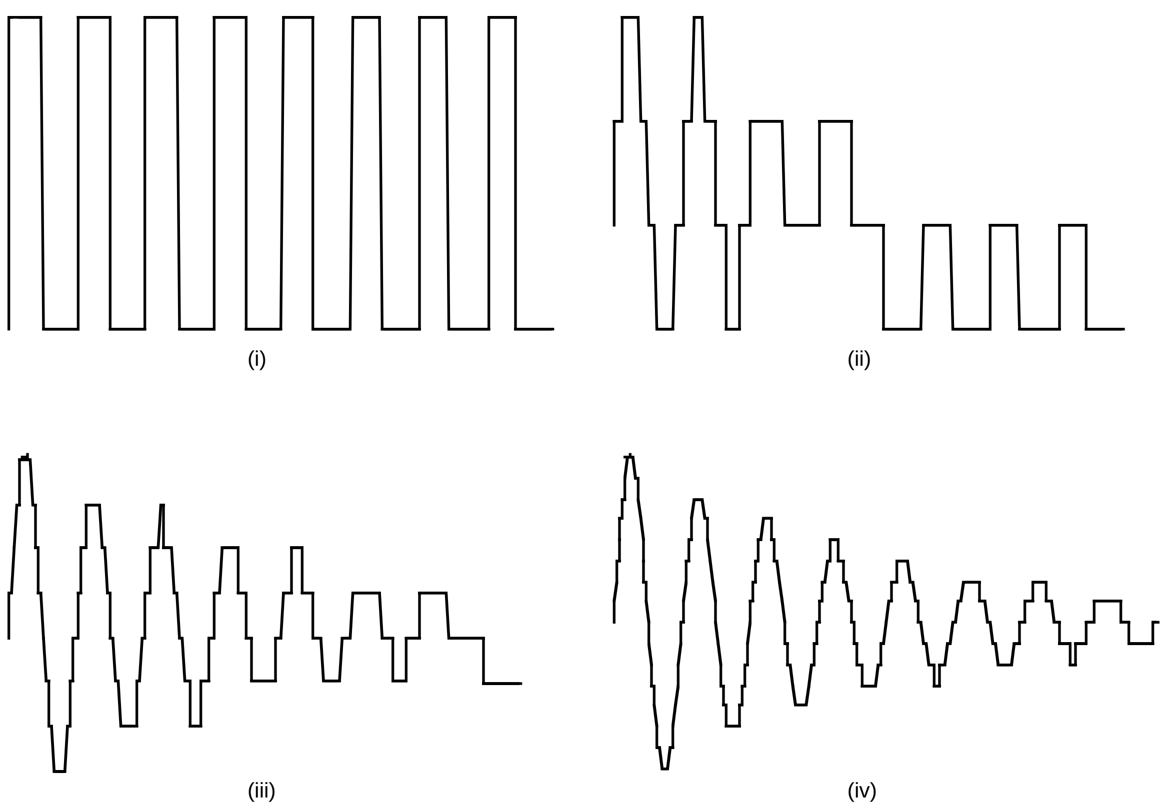


Figure 9 **A decaying sine wave represented with different word lengths: (i) 1-bit, (ii) 2-bit, (iii) 3-bit, (iv) 4-bit.**

Clearly a one-bit word gives a very crude approximation to the original signal while four bits is beginning to do a good job of reproducing the original decaying sine wave. When a good strong signal is being quantized (its word length is being shortened), quantization usually sounds like additive noise. But notice that as the signal decays in the above figures, fewer and fewer quantization levels are being exercised until, like the one bit example, there are only two levels being toggled. With just two levels, your signal has become a square wave.

Controlling the bit level of the quantizer is done with the DynamRange parameter (dynamic range). At **0 dB** we are at a one-bit word length. Every 6 dB adds approximately one bit, so at **144 dB**, the word length is 24 bits. The quantizer works by cutting the gain of the input signal, making the lowest bits fall off the end of the word. The signal is then boosted back up so we can hear it. At very low DynamRange settings, the step from one bit level to the next can become larger than the input signal. The signal can still make the quantizer toggle between bit level whenever the signal crosses the zero signal level, but with the larger bit levels, the output will get louder and louder. The Headroom parameter prevents this from happening. When the DynamRange parameter is lower than the Headroom parameter, no more signal boost is added to counter-act the cut used to quantize the signal. Find the DynamRange level at which the output starts to get too loud, then set Headroom to that level. You can then change the DynamRange value without worrying about changing the signal level. Headroom is a parameter that you set to match your signal level, then leave it alone.

At very low DynamRange values, the quantization becomes very sensitive to DC offset. It affects where your signal crosses the digital zero level. A DC offset adds a constant positive or negative level to the signal. By adding positive DC offset, the signal will tend to quantize more often to a higher bit level than to a lower bit level. In extreme cases (which is what we're looking for, after all), the quantized signal will sputter, as it is stuck at one level most of the time, but occasionally toggles to another level.

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

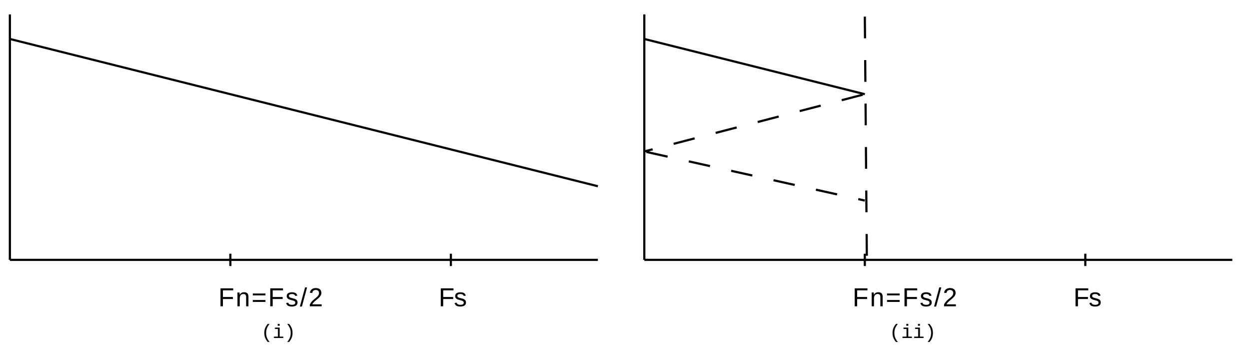


Figure 10 **Spectra of (i) an analog signal and the (ii) same signal after sampling without filtering**

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

Parameters:

Page 1

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

Page 2

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

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

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

DynamRange The digital dynamic range controls signal quantization, or how many bits to remove from the signal data words. At **0 dB** the hottest of signals will toggle between only two bit (or quantization) levels. Every 6 dB added doubles the number of quantization levels. If the signal has a lot of headroom (available signal level before digital clipping), then not all quantization levels will be reached.

Headroom When the signal has a lot of headroom (available signal level before digital clipping), turning down DynamRange can cause the amplitude of adjacent quantization levels to exceed the input signal level. This causes the output to get very loud. Set Headroom to match the amount of digital signal level still available (headroom). This is easily done by finding the DynamRange level at which the signal starts getting louder and matching Headroom to that value.

dc Offset Adds a positive DC offset to the input signal. By adding DC offset, you can alter the position where digital zero is with respect to you signal. At low DynamRange settings, adding DC offset can may the output sputter. dc Offset is expressed in decibels (dB) relative to full-scale digital.

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

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

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

745 Pitcher+MiniVerb

Combination algorithm of Pitcher followed by MiniVerb

PAUs: 2

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

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

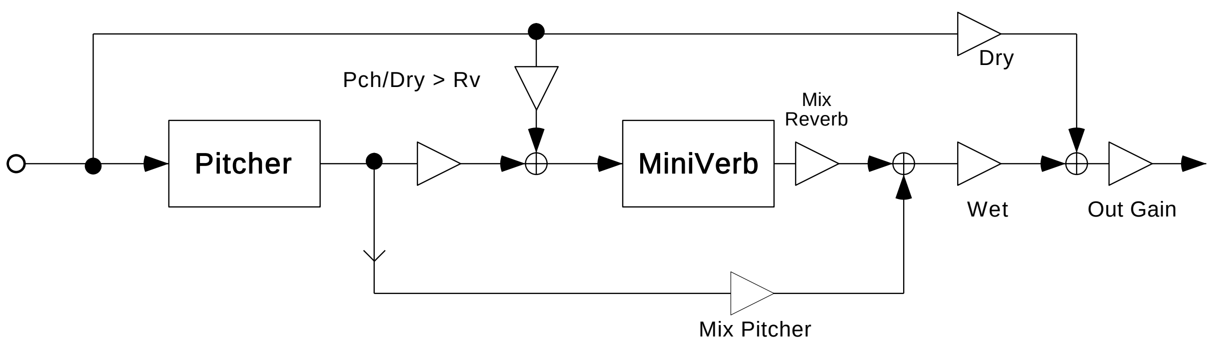


Figure 11 Signal flow of Pitcher+MiniVerb

Parameters:

Page 1

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

Page 2

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

Page 3

Pch/Dry>Rv	0 to 100 %		
Rv Type	Hall1, ...		
Rv Time	0.5 to 30.0 s, Inf		
Rv DiffScl	0.00 to 2.00x	Rv Density	0.00 to 4.00x
Rv SizeScl	0.00 to 4.00x	Rv HFDamp	8 to 25088 Hz
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms

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

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

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

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

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

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

Pt Odd Wts
Pt Pair Wts
Pt 1/4 Wts
Pt 1/2 Wts
These parameters control the exact shape of the frequency response of **Poly Pitcher** An exact description of what each one does is, unfortunately, impossible, since there is a great deal of interaction between them. For examples, examine the figures in the section on **Pitcher**.

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

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

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

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

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

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

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

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

746 Reverb+Compress

A reverb and compressor in series.

PAUs: 2

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

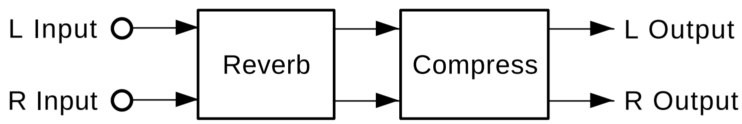


Figure 12 Simplified block diagrams of Reverb+Compress

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

The compressor reduces the signal level when the signal exceeds a threshold. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the sum of the magnitudes of the left and right channels. A compression ratio of **1:1** will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and **1:1**, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

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

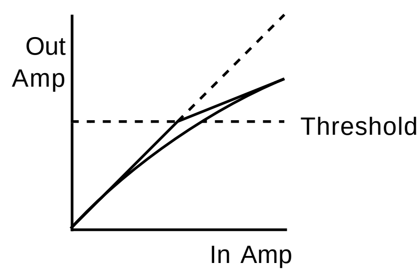


Figure 13 Soft-Knee compression characteristics

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may overshoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: CompSmooth. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to **Out**; for feedback compression, set it to **In**. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

In the feedback configuration, the signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens. In the feed-forward configuration, the delay affects both the main signal and the side chain, and so is of limited usefulness. In compressors which use more than 1 PAU, the delay affects the main signal only, regardless of the side chain configuration.

A meter displays the amount of gain reduction applied to the signal as a result of compression.

Parameters:

Page 1

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

Page 2

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

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

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

ReverbW/D This is a simple mix of the reverb input (dry) with the reverb output (wet) to produce the final reverb output.

ReverbGain An overall level control of the reverb's output (applied after the reverb Wet / Dry mix).

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

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

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

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

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

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

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

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

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

CompMakeUp Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

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

781 St Chorus+Delay
784 St Flange+Delay

Combination effect algorithms using time/frequency units instead of tempo

PAUs: 1 or 2

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

790 Gate+Cmp[EQ]+Vrb

Combination algorithm designed for vocal processing.

PAUs: 4 each

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

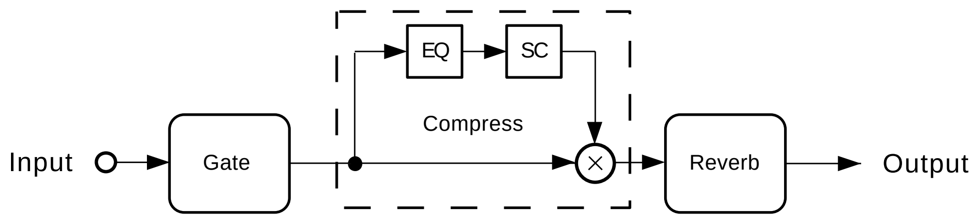

Figure 14 Gate+Cmp[EQ]+Vrb

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

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

Parameters:

Page 1

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

Page 2

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		Reduction	-dB 60 40 ❃ 16 ❃ 8 4 0

Page 3

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

Page 4

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

Page 5

Reverb W/D	0 to 100 %wet		
Rv Type	Hall1, etc.		
Rv Time	0.5 to 30.0 s, Inf		
Rv DiffScl	0.00 to 2.00x	Rv Density	0.00 to 4.00x
Rv SizeScl	0.00 to 4.00x	Rv HF Damp	8 to 25088 Hz
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms

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

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

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

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

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

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

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

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

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

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

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

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

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

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

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

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

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

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

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

CmpSCBassG, Bass Gain The amount of boost or cut that the bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

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

CmpSCTrebG, Treb Gain The amount of boost or cut that the treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

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

CmpSCMidG, Mid Gain The amount of boost or cut that the parametric mid filter should apply in dB to the specified frequency band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

CmpSCMidF, Mid Freq The center frequency of the parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

CmpSCMidW, Mid Width The bandwidth of the side chain parametric mid filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

Reverb W/D A simple mix of the reverb sound with the dry (compressed) sound.

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

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

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

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

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

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

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

792 Gate+TubeAmp

Combination algorithm designed for guitar processing.

PAUs: 3

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

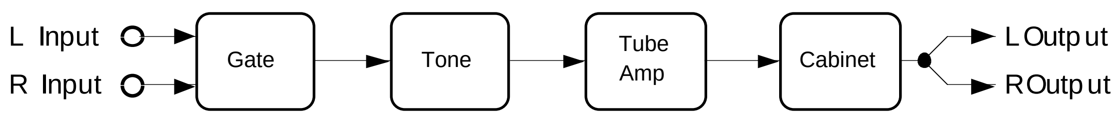

Figure 15 Gate+TubeAmp

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

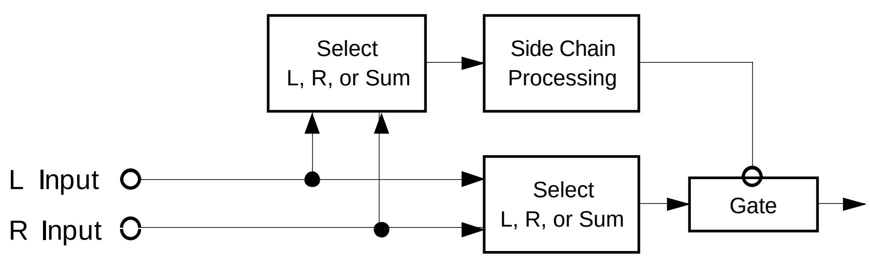

Figure 16 Gate routings

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

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

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

Parameters (Gate+TubeAmp):

Page 1

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

Page 2

<table>
<tr><td>Gate Thres</td><td>-79.0 to 0.0 dB</td><td>Gate Time</td><td>25 to 3000 ms</td></tr>
<tr><td>Gate Duck</td><td>On or Off</td><td>Gate Atk</td><td>0.0 to 228.0 ms</td></tr>
<tr><td></td><td></td><td>Gate Rel</td><td>0 to 3000 ms</td></tr>
<tr><td></td><td></td><td>GateSigDly</td><td>0.0 to 25.0 ms</td></tr>
<tr><td colspan="3">||||||||||||||||||||||||||||||</td><td>Reduction</td></tr>
<tr><td colspan="4">-dB 60 40 ✻ 16 ✻ 8 4 0</td></tr>
</table>

Page 3

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

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

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

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

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

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

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

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

Gate Time The time in seconds that the gate will stay fully on after the signal envelope rises above threshold. The gate timer is started or restarted whenever the signal envelope rises above threshold.

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

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

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

Bass Tone
Mid Tone
Treb Tone Adjusts the three bands of tone control integrated with the distortion drive circuit. Flattest response is obtained by setting Mid Tone to **10.0** and both Bass Tone and Treb Tone to 0.0.

Tube Drive Adjusts the gain into the distortion circuit. Higher values produce more distortion.

Cab Preset Eight preset cabinets have been created based on measurements of real guitar amplifier cabinets. The presets are **Basic**, **Lead 12**, **2x12**, **Open 12**, **Open 10**, **4x12**, **Hot 2x12**, and **Hot 12**.

Warmth Adjusts a 1 pole (6dB/oct) lowpass filter applied after distortion.

914 Revrse LaserVerb

A bizarre reverb which runs backwards in time.

PAUs: 4

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

Revrse LaserVerb is **LaserVerb** in reverse, so when it is fed an impulsive sound such as a snare drum, it plays the impulse back as a delayed train of closely spaced impulses, and as time passes, the spacing between the impulses gets closer until they coalesce at the “hit.” The close spacing of the impulses produces a discernible buzzy pitch which gets higher as the impulse spacing decreases. The following figure is a simplified representation of the **Revrse LaserVerb** impulse response. (An impulse response of a system is what you would see if you had an oscilloscope on the system output and you gave the system an impulse or a spike for an input.)

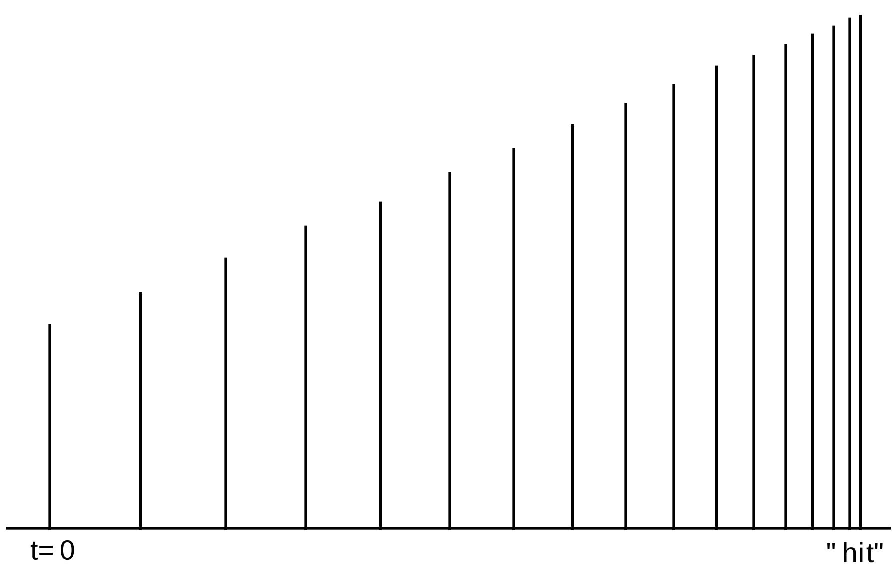

Figure 17 Simplified impulse response of Revrse LaserVerb

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

The Spacing parameter controls the initial separation of impulses in the impulse response and the rate of their subsequent separation. Low values result in a high initial pitch (impulses are more closely spaced) and takes longer for the pitch to lower.

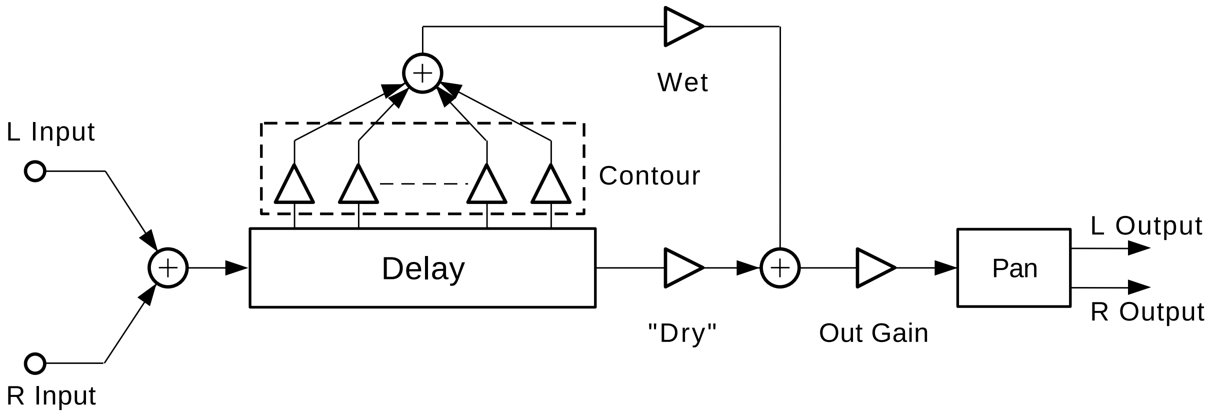

Figure 18 Revrse LaserVerb

Parameters:

Page 1

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

Page 2

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

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

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

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

Pan The left and right inputs get summed to mono, the mono signal passes through the **Revrse LaserVerb**, and the final mono output is panned to the left and right outputs. Panning ranges from **-100%** (fully left), through **0%** (centered), through to **100%** (fully right).

Dly Coarse You can set the overall delay length from **0** to **5** seconds. Lengthening the delay will increase the duration or decay time of the reverb.

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

Spacing Determines the starting pitch of the ascending buzz and how fast it ascends. The Spacing parameter sets the initial separation of impulses in the impulse response and subsequent rate of decreasing impulse separation. The spacing between impulses is given in samples and may be a fraction of a sample. (A sample is the time between successive digital words which is 20.8 µs or 1 / 48000 seconds.) For low values, the buzz builds to a higher frequency than for higher Spacing settings.

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

915 Gated LaserVerb

The LaserVerb algorithm with a gate on the output.

PAUs: 3

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

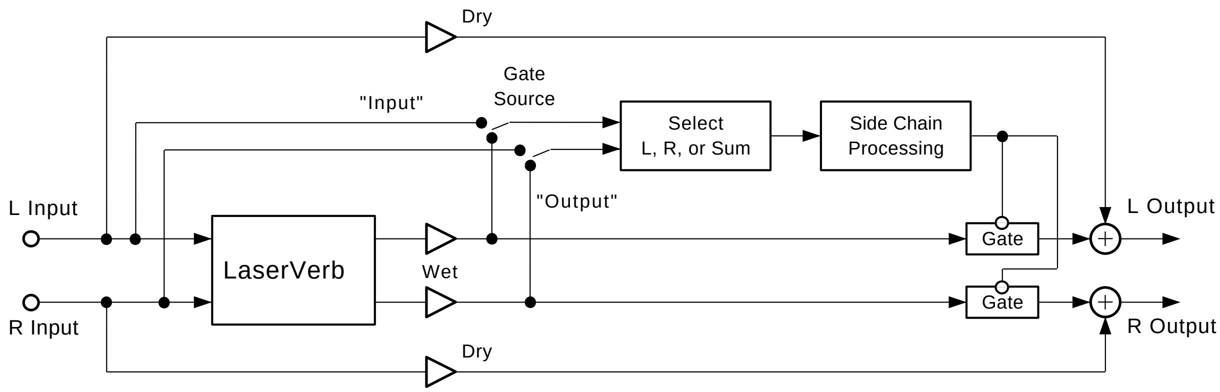

Figure 19 Signal flow of Gated LaserVerb

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

Parameters:

Page 1

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

Page 2

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

Page 3

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		||||||||||||||||||||||||||||	Reduction
	-dB 60 40 ❄ 16 ❄ 8 4		0

Wet/Dry The amount of reverbed and gated (wet) signal relative to unaffected (dry) signal. The gate is on the wet signal path.

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

Fdbk Lvl The percentage of the reverb output to feed back or return to the reverb input. Turning up the feedback is a way to stretch out the duration of the reverb, or, if the reverb is set to behave as a delay, to repeat the delay. The higher feedback is set, the longer the decay or echo will last.

Xcouple **LaserVerb Lite** is a stereo effect. The cross-coupling control lets you send the sum of the input and feedback from one channel to its own **LaserVerb** effect (0% cross coupling) or to the other channel's effect (100% cross coupling) or somewhere in between.

HF Damping The damping of high frequencies relative to low frequencies. When set to the highest frequency (**25088 Hz**), there is no damping and all frequencies decay at the same rate. At lower frequency settings, high frequency signal components will decay faster than low frequency components. If set too low, everything will decay almost immediately.

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

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

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

Dly Coarse You can set the overall delay length from **0** to **5** seconds. Lengthening the delay will increase the duration or decay time of the reverb. To reduce **LaserVerb** to a simple delay, set the Contour and Feedback controls to **0%**. Use a delay of about half a second as a starting point.

Dly Fine The delay fine adjust is added to the delay coarse adjust to provide a delay resolution down to **0.1 ms**.

Spacing Determines the starting pitch of the descending buzz and how fast it descends. The Spacing parameter sets the initial separation of impulses in the impulse response and subsequent rate of increasing impulse separation. The spacing between impulses is given in samples and may be a fraction of a sample. (A sample is the time between successive digital words which is 20.8 µs or 1 / 48000 seconds.) For low values, the buzz starts at high frequencies and drops slowly. At high values the buzz starts at a lower pitch and drops rapidly.

Contour Controls the overall envelope shape of the reverb. When set to a high value, sounds passed through the reverb start at a high level and slowly decay. As the control value is reduced, it takes some time for the effect to build up before decaying. At a value of around **34**, the reverb is behaving like a reverse reverb, building up to a hit. When the Contour is set to **0**, LaserVerb is reduced to a simple delay.

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

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

Gate Time The time in seconds that the gate will stay fully on after the signal envelope rises above threshold. The gate timer is started or restarted whenever the signal envelope rises above threshold.

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

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

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

916 Poly Pitcher

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

PAUs: 2

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

Parameters:

Page 1

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

Page 2

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

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

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

Odd Wts
Pair Wts
Quartr Wts
Half Wts These parameters control the exact shape of the frequency response of **Poly Pitcher**. An exact description of what each one does is, unfortunately, impossible, since there is a great deal of interaction between them. For examples, examine the figures in the section on **Pitcher**.

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

Pitch A, B The fundamental pitch imposed upon the input expressed in semitone scale intervals. Pitcher A and pitcher B may be set independently.

PchOff AL
PchOff AR
PchOff BL
PchOff BR An offset from the pitch frequency in semitones. Not only are the A and B pitchers treated separately, the left and right channels have their own controls for increased stereo separation. Pitch offset may be useful as a modifiable control resembling pitch bend.

917 Frequency Offset
918 MutualFreqOffset

Single Side Band Modulation

PAUs: 2

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

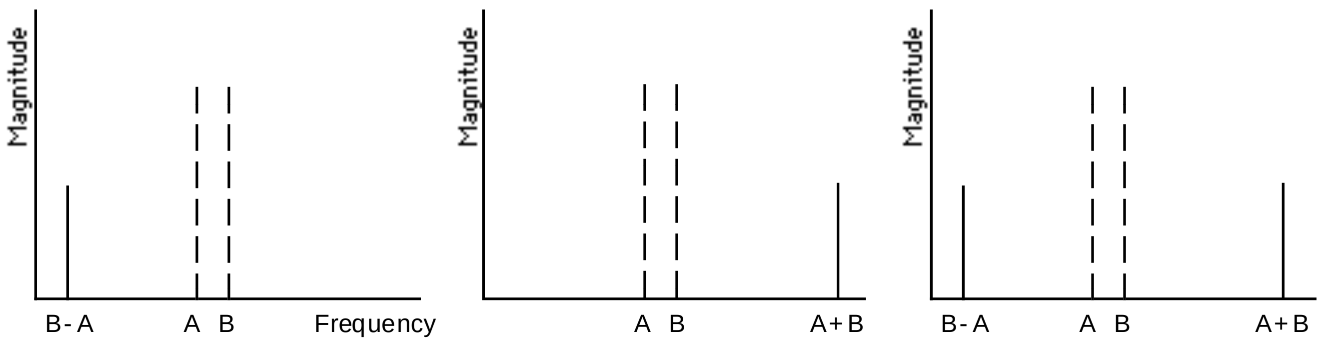

Figure 20 **Single side band modulation (frequency offset) (i) down and (ii) up. When combined (iii) we get dual side band modulation (ring modulation).**

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

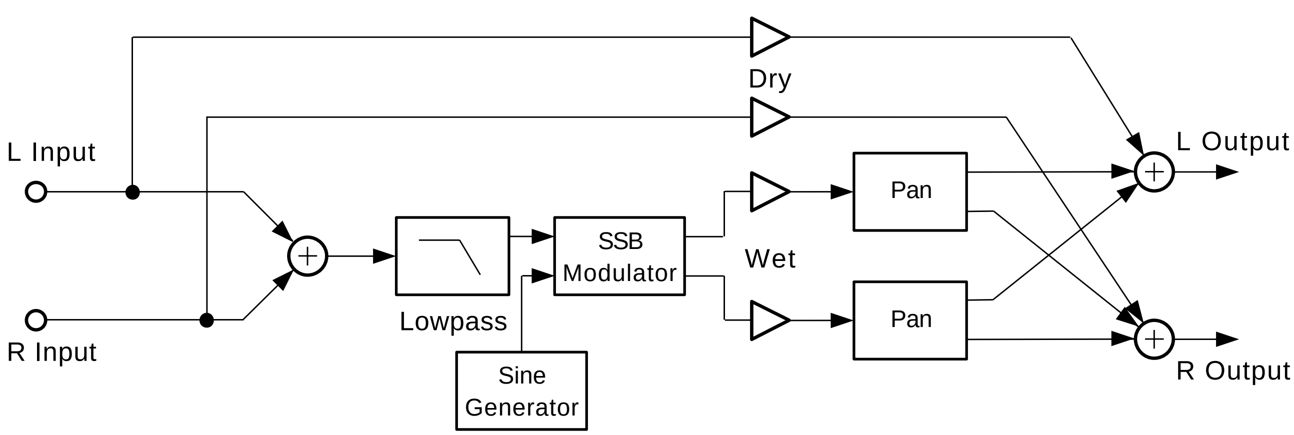

Figure 21 **Block diagram of Frequency Offset**

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

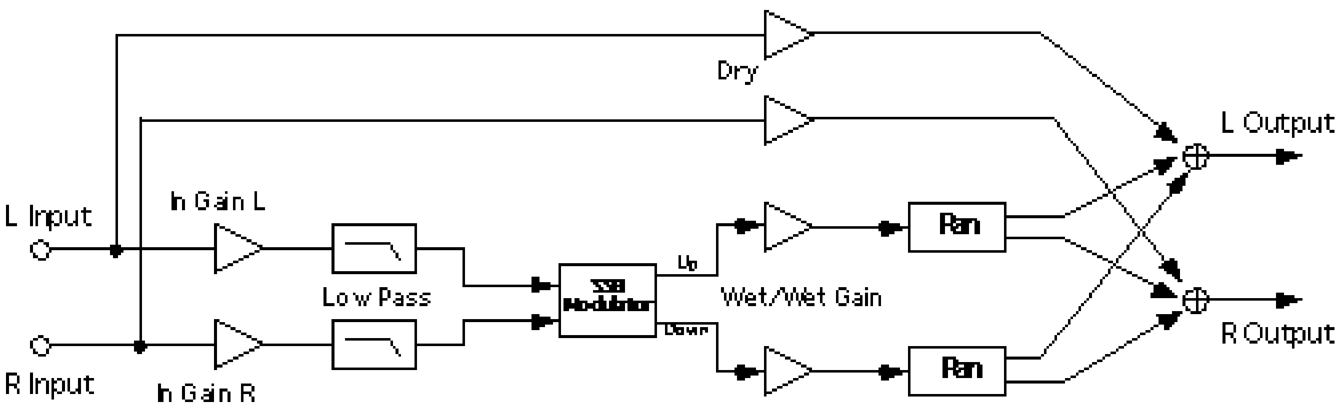

Figure 22 **Block diagram of MutualFreqOffset**

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

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

Parameters (Frequency Offset):

Page 1

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

Page 2

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

Parameters (MutualFreqOffset):

Page 1

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

Page 2

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

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

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

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

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

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

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

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

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

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

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

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

919 WackedPitchLFO

An LFO based pitch shifter.

PAUs: 3

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

Relatively.

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

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

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

Parameters:

Page 1

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

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

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

Feedback By introducing feedback, the pitch can be made to continually rise or fall as the signal makes successive passes through the pitch shifter.

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

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

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

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

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

920 Chaos!

Fun with chaos and instability

PAUs: 2

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

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

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

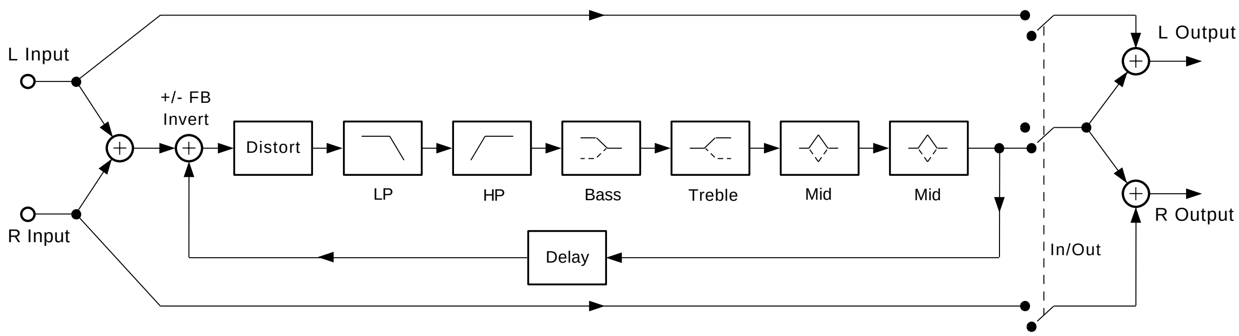

Figure 23 Chaos!

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

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

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

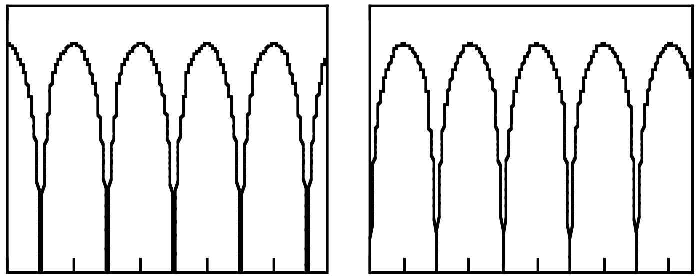

Figure 24 Resonating frequencies with FB Invert set to (i) Out and set to (ii) In.

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

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

Parameters:

Page 1

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

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

In/Out When set to **In**, the effect is active. When set to **Out**, the effect is bypassed.

Out Gain The overall gain or amplitude at the output of the effect. The output gain is outside and after the feedback loop.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

948 Band Compress

Stereo algorithm to compress a single frequency band

PAUs: 3

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

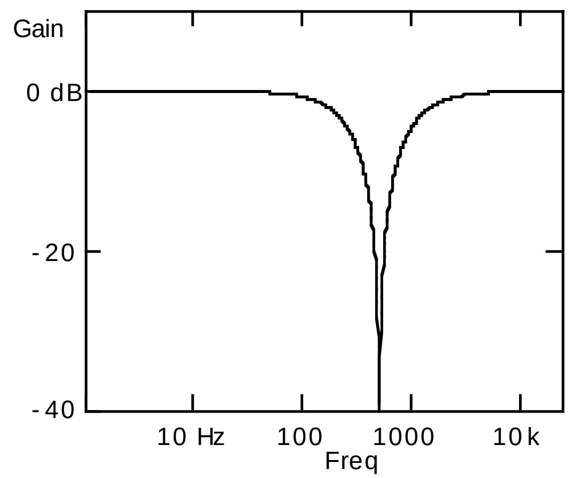

Figure 25 Band Compress filtering at full compression

The compressor reduces the signal level when the signal level exceeds a threshold. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the sum of the magnitudes of the left and right channels. A compression ratio of **1:1** will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and **1:1**, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

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

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

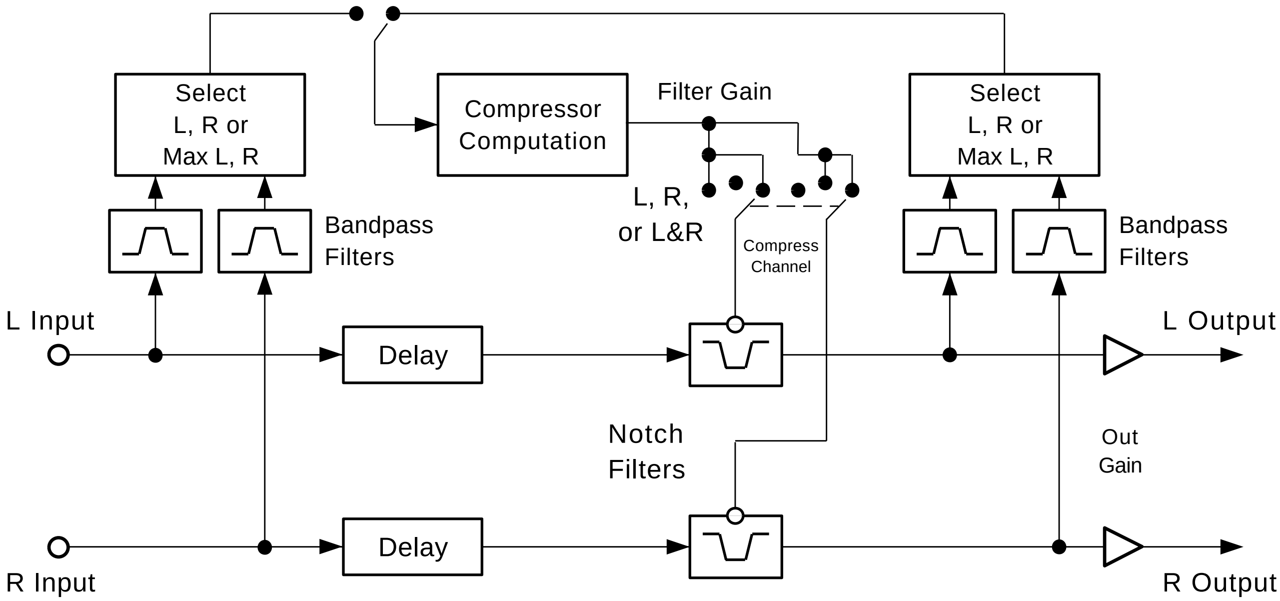

Figure 26   Band Compress block diagram

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

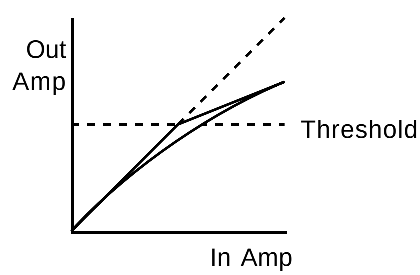

Figure 27   Soft-Knee compression characteristic

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may overshoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: SmoothTime. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to **Out**; for feedback compression, set it to **In**. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

In the feedback configuration, the signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing “knows” what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of compression.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Band FreqC	8 to 25088 Hz	FdbkCmprs	In or Out
Band FreqF	-100 to 100 ct	SC Input	L, R, L & R
Band Width	0.010 to 5.000 oct	ComprsChan	L, R, L & R

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1.0:1 to 100.0:1, Inf:1
Rel Time	0 to 3000 ms	Threshold	-79.0 to 0.0 dB
SmoothTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
Signal Dly	0.0 to 25.0 ms		
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

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

Out Gain The output gain parameter may be used to increase the gain by as much as 24 dB, or reduce the gain to nothing. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

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

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

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

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

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

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

Atk Time The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Rel Time The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

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

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

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

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

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

949 CompressDualTime

Compression with 2 release time constants

PAUs: 2

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

Like a normal compressor, this compressor reduces the gain of a signal as the signal increases above the compression threshold set with CompThres parameter. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the side chain input signal (left, right or larger of left and right magnitudes). A compression ratio of **1:1** will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and **1:1**, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may overshoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

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

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

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

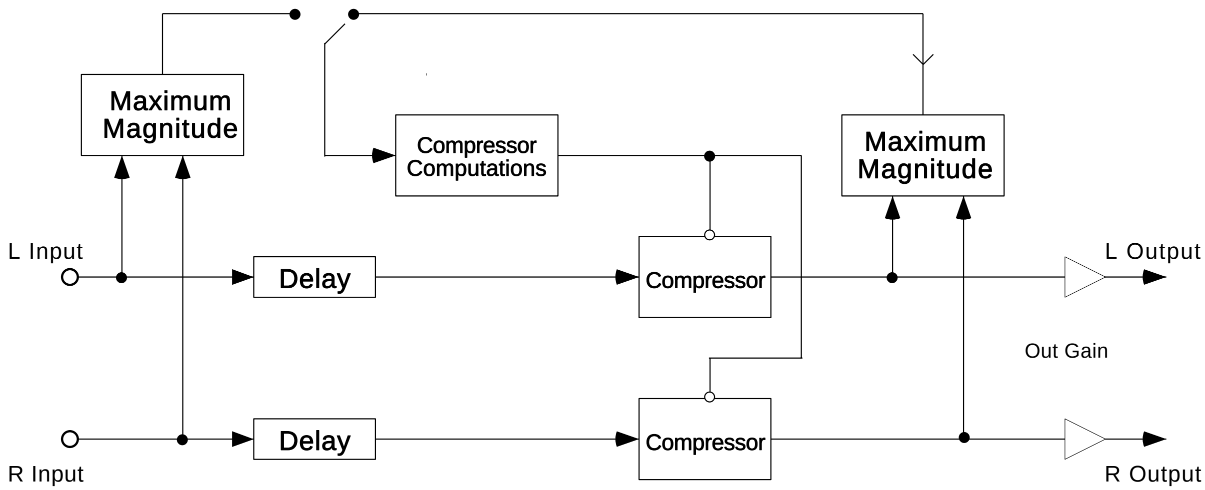

Figure 28 Opto Compress

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

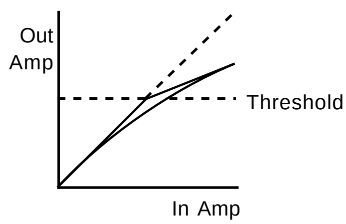

Figure 29 Soft-Knee compression characteristic

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: SmoothTime. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to **Out**; for feedback compression, set it to **In**. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

In the feedback configuration, the signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing “knows” what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of compression.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
FdbkComprs	In or Out	SC Input	L, R, L & R
Signal Dly	0.0 to 25.0 ms	ComprsChan	L, R, L & R

Page 2

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

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

Out Gain The output gain parameter may be used to increase the gain by as much as 24 dB, or reduce the gain to nothing. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

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

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

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

Atk Time The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

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

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

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

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

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

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

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

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

971 3 Band EQ

Bass and treble shelving filter and parametric EQs

PAUs: 1

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

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

Parameters:

Page 1 (3 Band EQ)

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

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

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

Bass Gain The amount of boost or cut that the filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

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

Treb Gain The amount of boost or cut that the filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

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

Mid Gain The amount of boost or cut that the filter should apply in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

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

972 HF Stimulate 1

High-frequency stimulator

PAUs: 1

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

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

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

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

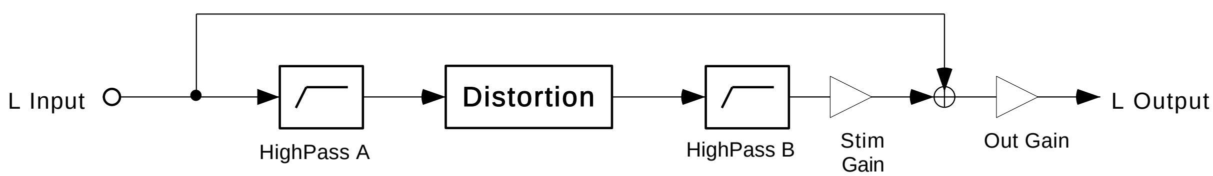

Figure 30 One channel of high-frequency stimulation

Parameters:

Page 1

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

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

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

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

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

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

975 HarmonicSuppress

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

PAUs: 2

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

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

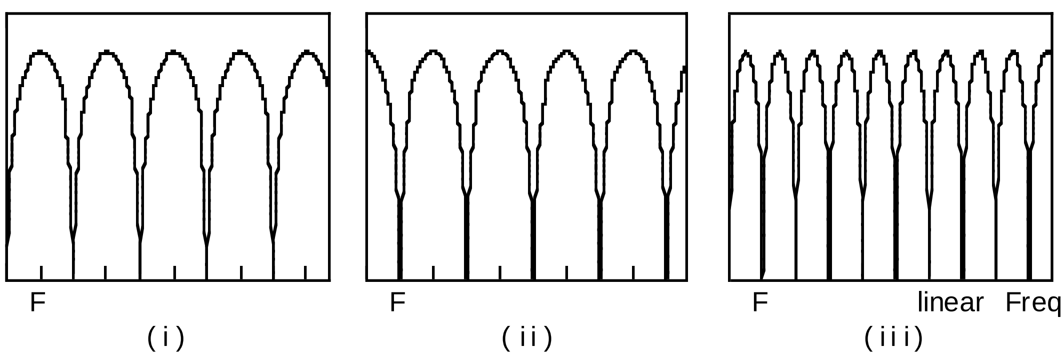

Figure 31 HarmonicSuppress filtering at full expansion
F marks fundamental
Harmonics are Even (i), Odd (ii), All (iii)

The algorithms expand the signal in the specified band(s) (reduce the signal's gain) when the signal falls below the expansion threshold in the specified band(s). You can select which channel, left (**L**), right (**R**) or the larger of the two (**L & R**) is used to control the expansion (side chain processing) with the SC Input parameter. You can also select which channel is actually expanded, again left (**L**), right (**R**) or both (**L & R**) using the ExpandChan parameter. The amount of expansion is expressed as an expansion ratio. Expanding a signal reduces its level below the threshold. The expansion ratio is the inverse of the slope of the expander input/output characteristic. An expansion ratio of **1:1** will have no effect on the signal. A zero ratio (**1:∞**), will expand all signal levels below the threshold level to the null or zero level. (This expander expands to **1:17** at most, but that's a lot.) Thresholds are expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

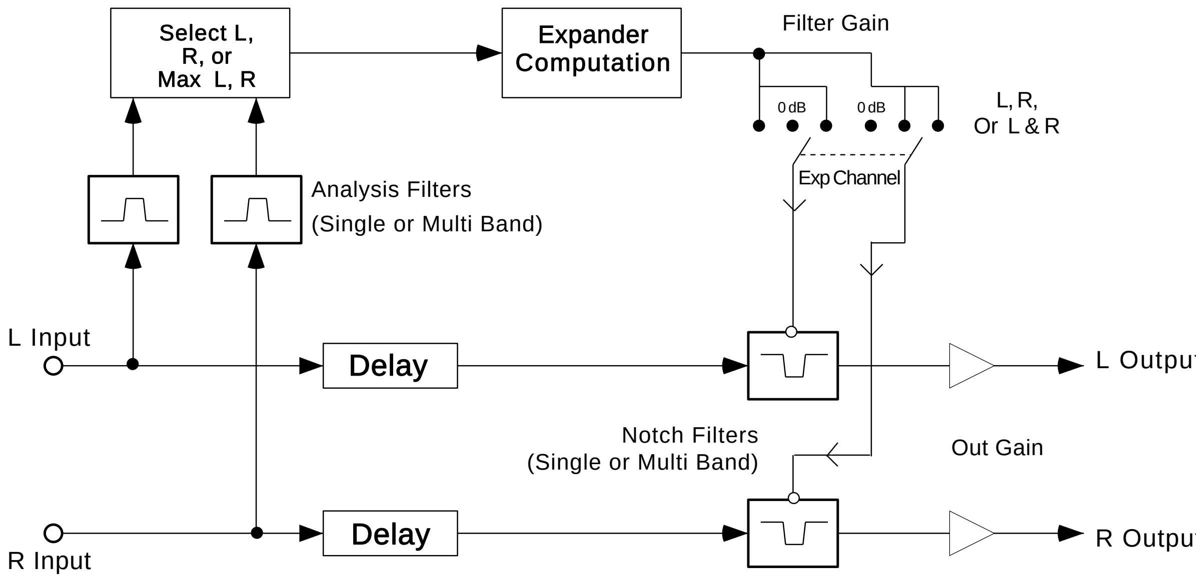

Figure 32 Band suppression

To determine how much to expand the signal, the expander must measure the signal level. Since musical signal levels will change over time, the expansion amounts must change as well. You can control how fast the expansion changes in response to changing signal levels with the attack and release time controls.

The attack time is defined as the time for the expansion to turn off when the signal rises above the threshold. This time should be very short for most applications. The expander release time is the time for the signal to expand down after the signal drops below threshold. The expander release time may be set quite long. An expander may be used to suppress background noise in the absence of signal, thus typical expander settings use a fast attack (to avoid losing real signal), slow release (to gradually fade out the noise), and the threshold set just above the noise level. You can set just how far to drop the noise with the expansion ratio.

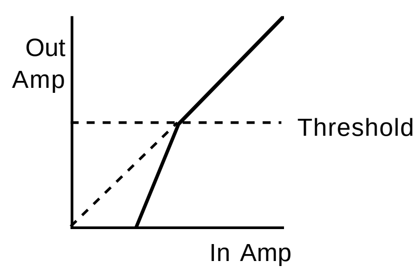

Figure 33 Expansion transfer characteristic

The signal being expanded may be delayed relative to the side chain processing. The delay allows the signal to stop being expanded just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by releasing the expander before the attack actually happens.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of expansion.

Parameters:

Page 1 (HarmonicSuppress)

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

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1:1.0 to 1:17.0
Rel Time	0 to 3000 ms	Threshold	-79.0 to 0.0 dB
SmoothTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
Signal Dly	0.0 to 25.0 ms		
		|||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

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

Out Gain The output gain parameter may be used to increase the gain by as much as 24 dB, or reduce the gain to nothing. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

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

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

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

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

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

Atk Time The time for the expander to increase the gain of the signal (turns off the expander) after the signal rises above threshold.

Rel Time The time for the expander to reduce the signal level when the signal drops below the threshold (turning on expansion).

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

Signal Dly The time in ms by which the input signal should be delayed with respect to expander side chain processing (i.e. side chain predelay). This allows the expansion to appear to turn off just before the signal actually rises.

Ratio The expansion ratio. High values (**1:17** max) are highly expanded, low values (**1:1** min) are moderately expanded.

Threshold The expansion threshold level in dBFS (decibels relative to full scale) below which the signal begins to be expanded.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to expansion.

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