

# Chapter 14

## Sampling and Sample Editing

### Setting Up For Sampling

Before you begin sampling, you'll need to connect the proper cables from your sample source to your K2600. The cables and input jacks you use depend on the sample format you choose, and the output configuration of your sample source.

Note that sampling requires the K2600 sampling option. Even without the sampling option, however, you still have access to all of the sample editing features covered later in this chapter, provided you have some sample RAM in your K2600. Samples can be loaded from disk, or dumped into the K2600 via MIDI Sample Dump Standard (SDS) or over SCSI using the SMDI protocol. See the *Musician's Reference* for information on the MIDI Sample Dump Standard and SMDI. Also see *SIMM Specifications* on page 9-2 of the *Musician's Reference* for information about sample RAM requirements.

#### Cables and Input Jacks

If you're going to be sampling from an analog source, you have two options:

- For unbalanced signals, use a 1/4-inch mono or stereo cable connected to the 1/4-inch (HiZ) stereo analog input jack
- For balanced signals, use balanced XLR (cannon) cables connected to one or both of the XLR mono analog inputs

Although it's possible to send a balanced signal on a 1/4-inch cable, avoid sending a balanced signal to the 1/4-inch jack when you're making stereo samples, since doing so can cause phase cancellation in your signals.

Using a mono cable sends the signal to the K2600's left channel. If you use a mono cable, be sure to set the Mode parameter on the SampleMode page to a value of **Mono(L)**.

If you're using a digital sample source, you can use either a coaxial cable or an optical cable, depending on the output format of your sample source. The coaxial input on the K2600 accepts a standard male XLR fitting. The K2600's XLR input is configured as follows: Pin 1 is Common, Pin 2 is High, and Pin 3 is Low. Depending on your sample source (a commercial DAT deck, for example), you may need to use a cable with an RCA connector on one end and an XLR connector on the other. In this case, you should tie the Common wire to the Low wire at the RCA end. Glyph Technologies (telephone 1-800-335-0345) carries an XLR-to-RCA cable that works with the K2600.

If your digital sample source has an optical output, connect your cable to the optical input jack above the coaxial jack on the K2600's rear panel. This jack is covered by a small plug which is easily removed. This plug should be kept in place whenever the optical input is not in use, since dust or dirt can cause the optical input to malfunction.

## Entering The Sampler

There are two different ways to get to the SampleMode page. The method you choose depends on the type of sampling you are doing—how many samples you are making and whether you need custom keymaps.

The difference between the two methods is primarily a matter of ease of access to the Keymap Editor. Once you have made your samples, you need to assign to a keymap and assign that keymap to a layer in a program. Refer to the section entitled *Building a Keymap* on page 14-39 for a step-by-step explanation of how to create keymaps.

### From Program, Setup, Master, or Quick Access Mode

The simplest way to get to the SampleMode page is from Program, Setup, Master, or Quick Access Mode. Press the soft button labelled **Sample** on any of these pages. This is a good method to use if you are making only a couple of samples, or if you want to assign each sample to its own keymap and program. Once you have created and saved your sample, you can press the **Preview** soft button. This button provides a quick way to create a program and keymap, with your sample assigned across the entire range of the keyboard. The program is a one-layer program that uses the settings from the Program **199 Default Program**.

### From the Keymap Editor

This is a better method to use if you're going to be doing lots of multi-sampling, or if you need to create custom keymaps in which you have your new samples assigned across the keyboard in one keymap. Call up Program **199 Default Program**. Press **Edit**, then **Keymap**. Select Keymap **168 Silence**, then press **Edit** again. This brings you to the Keymap Editor. (In fact you can choose any program and keymap you want to start with, but by choosing these, you are starting with a "blank slate.") Now from the Keymap Editor, press the **MIDI** mode button. This takes you to the SampleMode page. Once you have created and saved your samples, press **Exit**. You will now return to the Keymap-editor page, where you can immediately assign those samples across the keyboard. Once you have created and saved your keymap, you can either exit the Keymap Editor and create a program that uses your new keymap, or you can return to the SampleMode page for another round of sampling.

## Sampling Analog Signals

The K2600's analog sampling input is optimized for a low-impedance line level signal (-10 dBm). With a line-level signal, an input gain setting of 0 dB should prevent any clipping of the sample even at maximum output from the source. You can compensate for lower input levels with the Gain parameter on the SampleMode page.

If you're sampling through a microphone, you'll probably want to use a preamp to optimize your signal-to-noise ratio. If you don't have a preamp, you can adjust the Gain parameter on the SampleMode page. A setting of **21 dB** will give you reasonable results for many applications. This will increase the noise level as well, however.

Running your sample signal through a mixer before sending it to the K2600 will give you the most flexibility in controlling your signal level, since you can use its gain or pad if needed. This may add noise to the signal, however. For the cleanest possible signal, you'll want to connect your sample source directly to the K2600. The best results will be achieved by sampling from a digital source, using one of the K2600's digital sample inputs.

Assuming your connections are made, you're ready to set up your first sample recording. Select the SampleMode page (refer to *Entering The Sampler* above). The top line of the SampleMode page gives you the amount of free sample memory, and the amount of free program memory.

## Input

On the SampleMode page, you'll set the conditions for your sample recording. Depending on the input type you select, a different set of parameters will appear on this page. When you've selected analog input, the page appears as in the diagram below. The differences between analog and digital sampling are discussed in the section called *Sampling Digital Signals* on page 14-8.

```

SampleMode Samples:131072K Channel=2
Sample:None Src:Ext
Input :Analog Time:1s Mon:Off
Gain :0 dB
Rate :48.0KHz
Mode :Stereo
Thresh:Off -dB 60 40 * 16 * 8 4 0
Record Auto Timer Preview

```

The digital meters at the lower right of the display give a good indication of your sample level. When you send a signal from your sample source, you should see the meters respond.

### Src

The possible values for the Src parameter are Internal (**Int**) or External (**Ext**). Choose a value of **Ext** when you want to sample the signal from an external source that's connected to one of the K2600's sampling inputs. Use a value of **Int** if you want to sample the K2600's own output.

### Gain

The meters are calibrated in -dB units. A level of **0 dB** indicates the maximum signal without clipping. The sample will be free of clipping as long as the meter levels don't exceed 0 dB. For optimum results, you should adjust the K2600's Gain parameter (or the gain from your sample source) so that the signal stays below 0 dB. Otherwise, the signal will be clipped, causing the loss of sample data, and usually resulting in audible distortion of the resulting sample. A few clips (fewer than 100) may not cause any appreciable distortion. You'll get the best signal-to-noise ratio with meter levels as close to 0 dB as possible, although you'll find that samples with maximum meter readings as low as -12 dB can sound remarkably noise-free.

The relatively slow LCD output of the meter levels cannot show every peak in the incoming signal. Therefore, you won't necessarily see every transient in every sample you take. You will be able to see any transient that is clipped, however, since whenever a clip occurs, the K2600 will display the word "CLIP" above the meters, and will flash the Master-mode LED. It will also give you the number of clips in each sample before you save it.

### Rate

After you've set your levels, you need to select the sample rate. You have four rates to choose from. The tradeoffs that determine your best sampling rate are frequency response and storage requirements. Higher sample rates capture more frequency content from your samples, but take up more memory. Lower rates give you more sample time, but don't give the same frequency response as higher rates. Rates of 29.4 or 32 KHz yield a flat response up to about 14 and 15 KHz, respectively. 44.1 and 48 KHz yield a flat response up to 20 KHz, which is the upper limit of audibility for most humans. The lower rates may be adequate for most sounds, since many sounds have little content above 15 KHz. Sounds with a great deal of high-frequency content, such as cymbals, should probably be sampled at the higher rates. You can save memory by using lower sample rates for sounds without much high-frequency content—acoustic or electric bass, for example.

Another consideration in selecting sample rate is the K2600's transposition range during sample playback. The K2600 transposes samples by changing the sample playback rate; the higher the playback rate, the higher the pitch of the sample. The K2600 can achieve a maximum sample playback rate of 96 KHz. Normally, a sample made at 48 KHz can be transposed up a maximum of one octave, since the playback rate doubles for every octave of upward transposition. If you set the SmpSkp (sample skipping) parameter (on the KEYMAP page in the Program Editor) to **Auto** or **On**, you can transpose up two octaves at 48 KHz. A sample made at 29.4 KHz can be transposed up approximately 21 semitones (an octave and a sixth)—or 42 semitones with SmpSkp set to **Auto** or **On**. There is no limit on downward transposition, regardless of the sample rate. See page 6-24 for more information about sample skipping.

Each portion of a sample (each individual sample element made by the K2600 during the sampling process) takes up two bytes of sample memory. A one-second stereo sample at 48 KHz consists of 96,000 individual samples (48,000 x 2), taking up 192,000 bytes (about 188K) of sample memory. The same sample taken at 32 KHz takes up about 125K. A one-second mono sample taken at 32 KHz takes up about 63K.

If you plan to do a lot of sampling, you may want to consider adding sample memory to your K2600. SIMMS (Single In-line Memory Modules) are available at your dealer, or at most computer stores or mail-order houses. Be sure to read *Choosing and Installing SIMMs for K2600 Sample Memory* on page 9-2 of the *Musician's Reference* before you go SIMM shopping, though.

At a sampling rate of 44.1 KHz, each megabyte of sample RAM that you add increases your sample time by about 11.5 seconds (5.5 seconds for stereo samples). At 48KHz, each megabyte gives you about 10 seconds of mono sampling, and about 5 seconds of stereo sampling. Table 14-1 lists a few standard sample RAM configurations and their total sample time capacity (in seconds) at various sample rates.

Total RAM	Sampling Mode	Sampling Rate in KHz				Total Sampling Time (min:sec)
		29.4	32.0	44.1	48.0	
16M	Mono	4:20	4:16	3:04	2:48	
	Stereo	2:16	2:08	1:28	1:20	
32M	Mono	9:20	8:32	6:08	5:36	
	Stereo	4:32	4:16	2:56	2:40	
64M	Mono	18:40	17:04	12:16	11:12	
	Stereo	9:04	8:32	5:52	5:20	
128M	Mono	37:20	34:08	24:32	22:24	
	Stereo	18:08	17:04	11:44	10:40	

Table 14-1 RAM and Sampling Capacity

**Mode**

Use the Mode parameter to select mono or stereo sampling. (Keep in mind that stereo samples take up twice as much memory as mono samples.) Use a value of **Mono** for a mono signal. You can use either **Mono(L)** or **Mono(R)** to isolate either the left or right side of a stereo signal.

Audio sampling input doubles as a two channel "drum" trigger, allowing audio signals to trigger samples. On the SampleMode page, set Mode to **Trigger**. Adjust Thresh to control triggering sensitivity. This triggers the currently assigned click program. The left input will trigger click key note number +1, right input will trigger click key +2. The click key and click program can be accessed on the Song-mode MISC page.

There's also Live mode, which lets you connect any audio source to any of the K2600's sampling inputs (assuming you have the Sampling Option), and use that input just like a regular VAST program (the input goes through a DDP algorithm, then through KDFX, then to the audio outputs). Set Mode to LiveIn to use Live mode. See page 14-41 for more information about Live mode.

### Threshold (Thresh)

The Thresh parameter controls when the K2600 actually begins sampling incoming signals. If you set it to a value of **Off**, sampling begins immediately when you press the **Record** soft button. Otherwise the K2600 waits for the incoming signal to exceed a specified threshold before beginning to record. You can set the threshold from **-90** to **0 dB**, in 6 dB increments.

Sampler recording can also be triggered via the keyboard. Set Thresh to **Key**, then press **Auto**. Striking a MIDI note event now will trigger the sampler and assign the sample root to the key you struck, all in one easy step—making sample mapping easy and intuitive.

### Time

The Time parameter lets you determine how long the sample will be. The available sample time is a function of the sample rate and the amount of available sample memory. The K2600 calculates this automatically, and limits the maximum value of the Time parameter accordingly. At a value of **0** for this parameter, the K2600 will not record. (Of course, you can always stop sampling before the specified time by pressing the **Stop** soft button.)

### Sample

The Sample parameter lets you select any sample in memory for auditioning. This is a convenient way to listen to the samples you've made without having to create keymaps and programs for them manually. With a value of **None** for this parameter, the K2600 plays the last program or setup you selected before entering Sample mode. The list of values includes all ROM and RAM samples.

When you select a sample for auditioning, the K2600 automatically creates a temporary keymap and program, based on the settings for Program 199—which is a simple single-keymap program with few controller assignments—and the effects set to **0% wet (100% dry)**. Any edits you've made to Program 199 are reflected in the sample you audition. When you exit the SampleMode page, the temporary keymap and program disappear until the next time you audition a sample. You can create regular RAM keymaps and programs using the **Preview** soft button; see the discussion of the **Preview** button in the section called *Recording Samples* on page 14-6.

If you don't have enough free program RAM, you may be unable to audition samples, since the K2600 won't have enough RAM to create the temporary keymap and program. In this case, deleting a few objects from RAM will restore the audition feature.

### Monitor (Mon)

The Monitor parameter provides a convenient way to listen to what you're recording. When set to a value of **On**, any signal received at the analog sample input will appear at the K2600's Mix outputs and the headphone jack. Adjusting the input gain will affect the monitor gain as well. A clean monitor signal, however, does not guarantee a distortion-free sample. Always check the meters on the SampleMode page and look for the CLIP indicators to ensure that your sample is free of clipping. Note that the Mon parameter is not available when the Input parameter is set to a value of **Digital**. The Monitor feature applies only to the analog sampling inputs. You should monitor digital sources from the sources themselves.

## Recording Samples

Press the **Record** soft button to begin the sample recording process. If the **Thresh** parameter is set to a value of **Off**, recording will begin immediately, and will continue for the number of seconds indicated by the **Time** parameter. The display will indicate that recording is in process. Any other value for the **Thresh** parameter will cause the K2600 to wait until the specified threshold is exceeded, then recording will proceed normally. The display will indicate that you're making a threshold recording, but won't actually begin recording until the threshold is exceeded.

End the sampling process (either to save what you've done, or to abort) by pressing the **Stop** soft button.

When recording is complete, and you've pressed the **Stop** button, the K2600 will prompt you to strike a root key. The sample is assigned to the key you strike. This "root" is the key at which the sample will be played back without transposition. When sampling pitched sounds, it generally makes sense to assign a root key that matches the pitch of the original sample, although you can set the root key anywhere you like. If you press the **Default** soft button, the K2600 uses C 4. You can change the root key at any time on the **MISC** page in the Sample Editor.

When the root key has been assigned, the K2600 asks you if you want to save the sample. At this point the display will show one of two things—the number of clips, or if no clips occurred, the maximum level (in dB) of the sample signal.

You can listen to the sample before deciding whether to save it. If you decide not to keep the sample, press the **No** soft button, and you'll return to the **SampleMode** page. If you press **Yes**, you'll see the normal **Save** dialog. When you've saved the sample, you'll return to the **SampleMode** page. You'll also have the opportunity to name the sample. A recommended convention for naming samples is to include the root key as part of the name. This is particularly useful for pitched samples. Including the root key in the sample name helps when you are creating a keymap, because it tells you how much transposition of the sample you will hear depending on its key assignment.

Once the sample is recorded and saved, you may want to edit it, using the **TRIM** page, **LOOP** page, or any of the sample DSP functions.

### The Auto Soft Button

To save time when sampling with either the analog or digital inputs, you can use the **Auto** soft button. If the **Thresh** parameter is set to **Off**, sampling begins immediately. Once sampling is complete and you've pressed **Save**, the K2600 automatically assigns a root key of C 4, and saves the sample to the first available ID above 199.

If **Thresh** is set to a dB value, sampling begins when the incoming signal exceeds the **Thresh** level. If **Thresh** is set to **Key**, sampling begins when you strike a key.

Auto sampling is useful when you're making a series of samples that you expect to have the same approximate signal level. Since auto sampling doesn't show you the maximum signal level or the number of clips in the sample, it's a good idea to make your first sample in the series using the **Record** button. Once you have the input signal at the right level, you can make the rest of the samples in the series with fewer button presses.

### The Timer Soft Button

If you need to delay the beginning of your sample recording, you can press the **Timer** soft button instead of the **Record** or **Auto** soft buttons. This will begin a ten-second countdown before sample recording actually starts. The display will show the countdown. When the

countdown reaches zero, The Program, Setup, MIDI, and Master-mode LEDs will flash three times.

If you have the Thresh parameter set to a value of **Off**, sample recording begins immediately after the LEDs flash. If you have the Thresh parameter set to a dB value, sampling begins when the incoming signal exceeds the Thresh level. If Thresh is set to **Key**, sampling begins when you strike a key.

### The Preview Soft Button

When you've finished taking a sample, you can press the **Preview** soft button to automatically create a keymap and program using the new sample. It uses the settings for the Program **199 Default Program** as a template. Unlike the temporary keymap that's created when you audition a sample (and disappears when you select another sample), the preview keymap and program are stored in RAM and can be selected at a later time. The program and keymap will have the same name as the sample.

When you press the **Preview** soft button, the Bank dialog appears, prompting you to select a bank where the preview program will be stored. Select a bank, then press the **OK** soft button. The K2600 creates a keymap and a program, using the lowest available ID numbers in that bank for both the keymap and the program. The display tells you the ID of the new program.

### Multiple Sample Previews

The **Multi** soft button starts a process that lets you automatically build a program for previewing just about as many samples as you want (104 to be exact).

1. Press **Multi**. You'll see a list of available samples (keep in mind that these sample objects may consist of multiple sample roots). This list of sample objects is another version of the multiple object selector described on page 13-36.
2. Use the **Up/Down** cursor buttons and the **Select** soft button to highlight and select sample objects. The asterisk that appears indicates that the sample is selected. If you don't select any, the K2600 assumes you want to preview them all.
3. Press **OK**. If you've selected more than one sample, the K2600 asks you if you want to combine the sample objects into a single keymap and program. (If you've selected only one sample—one that doesn't consist of multiple sample roots—the K2600 returns to the Bank dialog, where you can select a different bank if you want, then press **OK**. The K2600 creates a keymap and program, tells you what the ID of the program is, and returns to the SampleMode page.)
4. At the "Combine into..." prompt, press **Yes**. The K2600 asks you if you want a tuned layout. (If you press **No**, at the "Combine into..." prompt, you return to the Bank dialog, where you can select a different bank if you want, then press **OK**. The K2600 creates a keymap and program for *each sample root*. If you're previewing a number of sample objects that consist of multiple sample roots, the list of programs can get quite long. In most cases it's much more convenient to combine the samples into one program.)
5. At the "Tuned Keymap Layout" prompt...decide how you want the sample objects to be laid out in the preview program. Pressing **Yes** maximizes the use of the keyboard. First you'll see the Bank dialog again. Press **OK**, and the K2600 takes the list of sample objects you selected in Step 2, and in order of their IDs, starts assigning them to their normal root keys. If two or more samples use the same root key(s), the most-recently assigned sample gets assigned to the next highest available key, and its coarse tune is adjusted so it plays at its root pitch. When all the roots are assigned to keys, the K2600 fills in between the roots,

so all keys play one of the samples. The number of keys playing each sample depends on the total number of sample roots you're previewing. Tuned layouts are useful for previewing pitched samples.

Pressing **No** at the "Tuned Keymap Layout" prompt is useful for previewing large numbers of samples, or percussion samples. When you press **No**, you'll see the Bank dialog. Press **OK**, and the K2600 takes the list of sample objects you selected in Step 2, and in order of their IDs, starts assigning them to keys, beginning at C 2, one root per key. The coarse tune gets adjusted so they all play at their root pitches. Keys below C 2 play the sample assigned to C 2, transposed accordingly. Above the highest key used, you'll hear the sample with the highest key assignment, transposed accordingly up to the upper transposition limit.

In either case, after the K2600 finishes processing the samples, it tells you the ID of the preview program (or the lowest ID if it created more than one program), then returns to the SampleMode page, with the preview program as the current program.

## Sampling the K2600's Output

You can sample the K2600's own sounds when in Analog sampling mode. To do so, set the Src parameter on the SampleMode page to a value of **Int**. Then, just press the **Record** soft button and start playing.

The K2600's "sample-while-play" capabilities offer a number of useful possibilities. It allows you, for example, to create composite sounds made up of several K2600 sounds or even sequences. This can help you make efficient use of the K2600's polyphony. By building composite sounds from other composite sounds, you could actually cause a frighteningly large number of K2600 sounds to become a single sample. The only constraints are your imagination—and the amount of sample RAM installed in your K2600.

You can also sample the K2600 directly into songs, using the RAM Tracks feature. See page 12-17.

## Sampling Digital Signals

The process for sampling through either of the digital inputs is essentially the same as that for sampling analog signals, although there are a few additional parameters associated with digital sampling formats.

You'll notice that the SampleMode page changes considerably when you change the value of the Input parameter from **Analog** to **Digital**. There are a few more settings to be made before you start recording. Note that the Src and Out parameters appear only if you have the both the sampling option and the digital I/O option. We'll discuss the Out parameter in more detail on page 19-3.

```
SampleMode Samples:1310r2K Channel=2
Sample:None Src:Ext
Input :Digital Time:1s Out:A/D
Cable :Coaxial
Format:AES/EBU
Mode :Stereo
Thresh:Off -dB 60 40 * 16 * 8 4 0
Record Auto Timer Preview
```



The first difference is the fact that there are no parameters for gain and sample rate. There's no need for a gain parameter because with digital sampling, since you're making an exact digital copy of the source signal. The Rate parameter is excluded because the K2600 automatically recognizes the source sample's rate and sets its own rate accordingly. Also, the Mon parameter does not appear when sampling digitally. Any monitoring you wish to do must be done from the sample source.

### Cable

Set the Cable parameter to a value of **Coaxial** or **Optical**, depending on the type of cable you're using. Many consumer products with digital output provide an optical jack. There is a small plug covering the K2600's optical input; this plug must be removed before you can connect the optical cable to the K2600. The plug should be replaced whenever the optical input is not in use.

If you plan to use a coaxial cable, select a value of **Coaxial** for the Cable parameter. If you're sampling from a consumer product like a CD player, you'll probably need to get an adaptor, or a cable with an RCA fitting on one end and an XLR (male) fitting on the K2600 end. Some professional products have an XLR fitting for their outputs, so an ordinary microphone cable will be suitable in most cases. Longer cables can cause signal loss, however, so if you need a long cable, you may need to get a special cable designed for digital information transfers. There's detailed information about cables for sampling on page 14-1.

### Format

Use the Format parameter to tell the K2600 the format of the incoming sample. Most consumer products use SPDIF (Sony/Philips Digital Interface Format), while most professional machines use the AES/EBU (Audio Engineering Society/European Broadcast Union) format. Refer to the owner's manual of your sample source for information regarding its digital format.

The Mode, Time, and Thresh parameters function for digital sampling just as they do for analog sampling.

If the K2600 detects an incoming clock signal, the display shows LOCK, and the sample rate of the signal. If you don't see LOCK, you're not getting signal, and you won't be able to sample. The K2600 automatically sets itself to the clock rate it detects. (For sample rates other than 48, 44.1 and 32, the rate doesn't show in the display, but the K2600 still samples the input correctly. You might need to adjust the coarse tune to get the proper root pitch.)

### Src

When Input is set to **Digital** on the SampleMode page, the Src parameter appears only if you also have the digital I/O option. It's the same parameter that appears when Input is **Analog**. Use a value of **Int** for Src when you want to sample the K2600's Mix output. Use **Ext** when you want to sample an external source via the coaxial or optical stereo digital input jacks.

If you have the digital I/O option, Src has another value: Return (**Rtn**). This enables you to sample a two-channel return from KDS Out A (this requires a Kurzweil DMTi or other device that's compatible with KDS digital format).

### Out

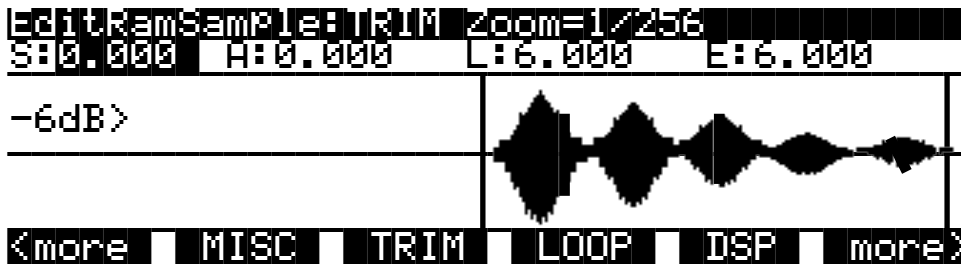
This parameter appears only if you also have the digital I/O option. It doesn't really have anything to do with sampling; it determines how the K2600 handles its digital audio output. See Chapter 19 for information about using the digital audio outputs.

## Editing Samples

Most of the functions within the Sample Editor follow a general pattern. There are two ways to enter the Sample Editor. If you start from Master mode and press the **Sample** soft button, then select a sample and press **Edit**, you can hear the isolated sample. If you want to hear the sample in the context of a program, enter the Sample Editor through the Program Editor: start by selecting a program in Program mode—usually the program containing the sample you want to edit. Press the **Edit** button to enter the Program Editor. Press the **KEYMAP** soft button to view the KEYMAP page. The KeyMap parameter is selected (highlighted) when the page appears. Press the **Edit** button to enter the Keymap Editor. The KeyRange parameter is selected when the page appears. The notes within the currently selected key range are the only ones that will be affected by your edits. You can hold the **Enter** button and trigger notes to select different key ranges.

If you want to select a different sample, use the cursor buttons to select the Sample parameter. Use the Alpha Wheel to select a sample. Press the **Edit** button once more, and you'll enter the Sample Editor. (Pressing the **Edit** button while in the Keymap Editor will enter the Sample Editor regardless of which parameter is selected.) The effects of the current program will be applied to the sample.

The TRIM page appears when you enter the Sample Editor. A representative TRIM page is shown below.



There are three basic sample editing pages—TRIM, LOOP, and MISC (Miscellaneous). The soft buttons for these pages are visible when you enter the Sample Editor. The **DSP** soft button is visible as well if you're editing a RAM sample; pressing it will take you to the DSP function page, where you can select a DSP function with the Alpha Wheel or **Plus/Minus** buttons.

The **DSP** soft button does not appear if you're viewing a ROM sample. Instead you'll see a **Link** soft button.

The **<more>** soft buttons will take you to the soft buttons for the other functions. You can trigger notes at any time while you're editing, to hear your changes as you make them.

### The Function Soft Buttons in the Sample Editor

In addition to the **MISC**, **TRIM**, and **LOOP** soft buttons, which select Sample-editor pages, there are several function soft buttons. As with other K2600 editors, the function soft buttons are labeled with upper and lower case letters, to distinguish them from the page selection soft buttons, which are labeled in all capital letters. The **<more>** soft buttons give you access to the other soft buttons that are available.

### Zoom- and Zoom+

These buttons are active only when you're viewing the TRIM and LOOP pages. They increase or decrease the resolution of the waveform display, enabling you to see a larger or smaller segment of the waveform of the currently selected sample. The top line of the display indicates the zoom position in terms of a fraction—for example,  $1/256$ —which indicates the number of individual sample elements represented by each display pixel. A value of  $1/256$  means that each pixel represents 256 individual sample elements. The maximum zoom setting of 1 shows you a very small segment of the sample. The minimum setting of  $1/16384$  shows you the largest possible segment of the sample. Each press of a **Zoom** soft button increases or decreases the zoom by a factor of 4.

As a convenience, the **Program** and **Setup** mode buttons also serve as zoom buttons while in the Sample Editor. You can press the two left soft buttons together to toggle between the default zoom setting and your current zoom setting.

### Gain- and Gain+

Also active only for the TRIM and LOOP pages, these buttons increase or decrease the magnification of the currently displayed sample waveform, enabling you to see the waveform in greater or lesser detail. At the left of the display, you'll see the magnification setting, which is expressed in dB units. You can adjust the magnification from -72 dB (maximum magnification) to 0 dB. This doesn't affect the actual amplitude of the sample, only the magnification of its display.

As a convenience, the **MIDI** and **Master** mode buttons also serve as gain adjustment buttons while in the Sample Editor.

The simplest way to think of the **Zoom** and **Gain** buttons is to remember that the **Zoom** buttons control the left/right magnification of the waveform, while the **Gain** buttons control the up/down magnification. Neither button has any affect on the sound of the sample. You'll often use the **Zoom** and **Gain** soft buttons together to focus in on a particular sample segment, then magnify it to see it in close detail.

For example, you might want to zoom out to view an entire sample waveform, to decide which segment you want to edit. You could then zoom in to focus on a particular segment. Once you've zoomed in, you may want to boost the Gain to enable you to set a new Start (S) point with greater precision, or ensure that you get a smooth loop transition.

### Abort

Use the **Abort** soft button to cancel a sample dump before it's complete. You'll be prompted to verify whether you really want to cancel the dump.

### Split

The **Split** soft button enables you to create two mono samples from a single stereo sample, or to split up a multi-root block of samples. When you press this button, the K2600 will prompt you: Split this sample? When you press the **Yes** soft button, you'll be prompted to enter the ID for the first sample. Select an ID with the Alpha Wheel or **Plus/Minus** buttons, then press the **OK** soft button. If you've selected a stereo sample, the K2600 splits it into left and right sides. If it's a block of samples, the K2600 splits it into individual sample roots. In either case, the split samples will automatically be assigned IDs, starting with the ID you select.

Splitting stereo samples enables you to use the separate sides individually, or to phase the samples by assigning each side to a separate keymap, then delaying one of the layers slightly.

#### Join

You can create sample objects that contain multiple sample roots. Many of the ROM samples are like this—multiple samples stored in memory as a single sample object. Joining samples is a great way to cut down on the number of IDs that you use for your samples, since joined samples all use the same object ID.

1. Press the **Split** soft button, then press the **Join** soft button. A list of RAM samples appears. This is another version of the multiple object selector, as described on page 13-36.
2. Use the **Up/Down** cursor buttons and the **Select** soft button to highlight and select samples (don't mix mono and stereo samples). The asterisk that appears indicates that the sample is selected.
3. Press **OK**. If the K2600 asks you "Are you sure?" press **Yes**. You'll return to the Sample Editor.
4. Save the sample. We recommend using a new ID. Every sample root in the joined sample uses this ID. The names and note numbers are different. When you save the sample, the K2600 will ask you if you want to copy the sample data. There's no need to do this; it will only take up additional memory. Everything works properly if you don't copy the data, and when you save your new sample to disk, everything gets saved accordingly.

When you return to the SampleMode page (or if you're looking at the Sample parameter in the Keymap Editor), you can scroll through the list of samples, and see your newly-created sample block. Notice the names and/or note numbers changing, while the ID remains the same throughout each sample block.

#### *A Few Notes About Joined Samples*

Don't try to join stereo and mono samples; the K2600 can join samples only if they're the same type.

You can't use any of the Sample Editor's DSP functions on joined samples; make sure to process each sample before joining.

#### Units

With the **Units** soft button you can change the units used to display the locations of the current sample's Start, Alt, Loop and End points. The default setting displays these points in seconds, that is, the number of seconds from the physical start of the sample. Pressing the **Units** soft button will change the units to samples—that is, the number of individual sample elements from the physical start of the sample. Press it again to return to a view of the sample in seconds.

As a convenience, the **Quick Access** mode button also serves as a units button while in the Sample Editor.

#### Link

The **Link** soft button lets you fix the interval between the Start, Alt, Loop and end points of the current sample, so it remains constant when you move one or more of the points. This is done by selecting the desired parameter with the cursor buttons, then pressing the **Link** soft button. The colon (:) following the parameter's name will change to an arrow (>) to indicate that it is linked. You can link any or all of the four sample points. When sample points are linked, moving one of them will move the linked points correspondingly. For example, suppose the current sample's Start (S) point is 0.0 seconds, and its Alt (A) point is 0.5 seconds. The interval between the sample's Start and Alt points is exactly half a second. If you select the Start parameter, then press the **Link** soft button, the Start point will be linked. This won't have any effect until you link at

least one more point. If you select the Alt parameter and press the **Link** soft button, the Start and Alt points will be linked. Now if you move the Start point to 1.0 seconds, the Alt point will automatically move to 1.5 seconds, preserving the half-second interval between Start and Alt. To remove the link on any of the points, select the point again, and press the **Link** soft button again. The arrow will change to a colon, indicating that the link has been removed.

When editing RAM samples, you can't assign Start, Alt, Loop, or End points that extend beyond the boundaries of the actual sample data (you *can* do this with ROM samples). Consequently, if you attempt to move any of the linked points of a RAM sample beyond the physical start or end of the sample, the K2600 changes the interval between the points accordingly. For example, suppose you have a sample with Start and Alt linked. Start is at 0 seconds, and Alt is at 1 second. If you move Alt to a later point, Start tracks with it, so the interval between Start and Alt remains at 1 second. If you move Alt to .5 seconds, the K2600 tries to move Start accordingly, but can't move it earlier than 0 seconds, so the interval between Start and Alt shrinks to .5 seconds. Even if you move Alt back to 1 second, the interval remains at .5 seconds. You'd have to reset Start manually to restore the original 1-second interval.

As a convenience, the **Song** mode button also serves as a link button while in the Sample Editor.

### Name, Save, Delete, and Dump

These soft buttons are similar to the **Name**, **Save**, **Delete**, and **Dump** soft buttons in the other editors, initiating the corresponding dialogs to name, save, delete, or dump the currently selected sample.

When you press the **Save** soft button, and choose to save the sample to a new ID, the K2600 will ask you if you want to copy the sample data. If you answer Yes, the K2600 will make a separate copy of the sample. If you answer No, the K2600 will simply mark the location of the original sample data and share the sample between the original and the edited sample. This can save a great deal of memory space. If you delete a sample that's partially or completely shared with another, the K2600 deletes only the portions that are unused by the shared sample, always optimizing its memory for maximum storage capacity.

Note that if you use the Utility function to view the objects currently stored in the K2600, you'll see each object listed separately, including shared samples. The shared samples will each indicate their size, even though they're referring to the same memory location. This might lead you to believe that you're using more memory for samples than you actually are. If you use the Utility function to calculate your total sample memory usage, remember not to include any shared samples in the total.

**Dump** lets you transfer (dump) samples using the MIDI sample dump protocol. See Chapter 6 of the *Musician's Reference* for more information.

## The Page Buttons

The soft buttons labeled in all capital letters select the various pages in the Sample Editor: **MISC**, **TRIM**, and **LOOP** (as well as **DSP** when you're editing RAM samples).

## The Miscellaneous (MISC) Page

On the MISC page, you'll set several parameters that affect the behavior of the current sample. These parameters affect the entire sample. The diagram of the MISC page shows a ROM sample, and since the DSP functions cannot be applied to ROM samples, the **DSP** soft button is not available. In its place is the **Link** soft button, which enables you to maintain equal times between various points in the sample—Start and Alt, for example.

The default values shown in this diagram reflect the settings for the Default program 199.

```

EditRomSample: MISC
RootKeyNum   : C 4      LoopSwitch: On
PitchAdjust  : 1ct      Playback   : Normal
VolumeAdjust : 0.0dB    AltSense   : Norm
AltVolAdjust : 0.0dB    IgnRelease : Off
DecayRate    : 950dB/   SampleSize : 0Kb
ReleaseRate  : 950dB/   SampleRate : 1000Hz
<more> MISC TRIM LOOP Link more>
    
```

Parameter	Range of Values
Root Key Number	C -1 to G 9
Pitch Adjust	Variable (depends on sample rate)
Volume Adjust	-64.0 to 63.5 dB
Alternative Volume Adjust	-64.0 to 63.5 dB
Decay Rate	0 to 5000 dB per second
Release Rate	0 to 5000 dB per second
Loop Switch	Off, On
Playback Mode	Normal, Reverse, Bidirectional
Alternative Sample Sense	Normal, Reverse
Ignore Release	Off, On

**Root Key Number**

The root key number represents the key at which the sample will play back without transposition (that is, at the same pitch as the pitch of the original sample). When you’re creating your own samples, the key you strike as the root key will be the key you see as the value for this parameter.

**Pitch Adjust**

Use this parameter to change the pitch of the sample relative to the key from which it’s played. Setting a value of **100cts**, for example, will cause the sample to play back one semitone higher than normal. This parameter is handy for fine tuning samples to each other if they’re slightly out of tune.

**Volume Adjust**

Uniformly boost or cut the amplitude of the entire sample. Compare this to the DSP Volume Adjust parameter, which lets you boost or cut the amplitude of a specified segment of a RAM sample.

**Alternative Start Volume Adjust (AltVolAdjust)**

This parameter sets the amplitude of the sample when the alternative start is used. See page 6-26 for a discussion of AltSwitch.

### Decay Rate

This parameter defines how long the sample takes to decay (fade) to zero amplitude (silence). Decay Rate affects each sample individually, and is in effect only when the amplitude envelope for the program (the Mode parameter on the AMPENV page in the Program Editor) is set to **Natural**. If Mode is **User**, the settings on the AMPENV page override the setting for DecayRate.

DecayRate operates in two different ways, depending on the nature of the sample. If it's a ROM sample or a Kurzweil-format RAM sample, DecayRate takes effect in the loop portion of the sample, after all the attack stages of the amplitude envelope are complete. For non-Kurzweil RAM samples, DecayRate takes effect as soon as the sample starts, regardless of the length of the attack stages.

### Release Rate

The release rate determines how long the sample will take to decay to zero amplitude when the note trigger is released. The higher the value, the faster the release rate. This release affects each sample individually, and is in effect only when the amplitude envelope for the program (the Mode parameter on the AMPENV page in the Program Editor) is set to **Natural**. In this case, the release begins as soon as the note is released. If Mode is **User**, the settings on the AMPENV page override the setting for ReleaseRate.

To create an extended sample loop that will play data after the sample's loop on key-up, set the Alt sample pointer after the sample end pointer, then set a relatively low value for the release rate.

### Loop Switch

This parameter activates or deactivates the looping of the currently selected sample. When set to **On**, the sample will loop according to the settings on the LOOP page. When set to **Off**, the sample will play through to its End point and stop.

### Playback Mode (Playback)

This parameter lets you modify the direction in which the sample is played. Set it to a value of **Reverse** if you want the sample to play from its End (E) point to its Start (S) point. Choose a value of **Bidirectional** to cause the sample to play from Start to End, then reverse direction and play again from End to Loop and back, repeating until the note trigger is released (this works only when the Loop Switch parameter is set to **On**).

### Alternative Sample Sense (AltSense)

This provides a convenient way to activate the alternative start of a sample. When set to **Normal**, the alternative start will be used when the Alt Switch control is **On** (this is set on the KEYMAP page), or when the control source assigned to it is above its midpoint. When set to **Reverse**, the alternative start will be used when the Alt Switch control is **Off**, or when the control source assigned to it is below its midpoint.

### Ignore Release (IgnRelease)

When set to a value of **Off**, the sample will release normally when the note trigger is released. When set to **On**, the note will not release, even when the note trigger is released. This setting should be used only with samples that normally decay to silence; nondecaying samples will play forever at this setting. This parameter is equivalent to the IgnRelease parameter on the LAYER page, but affects only the currently selected sample.

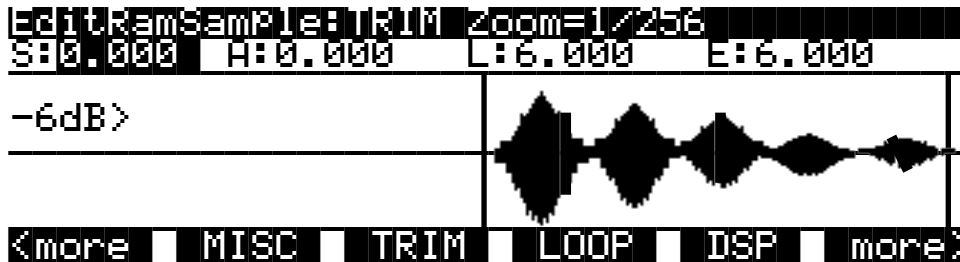
### Sample Size and Sample Rate

These are not editing parameters. They're on this page to give you a convenient place to check the size in kilobytes of the current sample, and the rate at which it was sampled.

## The TRIM Page

The TRIM page lets you set the Start, Alt, Loop, and End points of the current sample. The top line tells you whether you're editing a ROM or RAM sample, and indicates the zoom setting. At the left of the display is the Gain (display magnification) setting. This Gain setting doesn't affect the amplitude of the sample, just the view in the display.

In the diagram below, the sample points are expressed in individual sample elements. Pressing the **Units** soft button (or the **Quick Access** mode button) will display them in seconds.



The four parameters on this page are Start (S), Alternative Start (A), Loop (L), and End (E). Selecting these parameters and adjusting their values enables you to modify how the sample plays back when notes are triggered.

There are four vertical lines that indicate the settings of the four parameters. You'll see all four lines only if the values for each of the four parameters are different; otherwise, the lines overlap. Selecting one of the parameters moves the line corresponding to that parameter to the center of the display, where it flashes to indicate the parameter's position. Turning the Alpha Wheel will move the sample waveform relative to the line. The line remains in the center of the display, and the waveform shifts to indicate the new position of the point. You can also use the alphanumeric buttonpad to enter new values directly. Press the **Enter** button to register the values you enter.

The Start (S) point determines the beginning of the current sample. You can truncate the beginning of the sample by increasing the value of the Start (S) parameter. You might do this to remove silence at the beginning of a sample, or to remove some or all of the attack. You can't decrease the Start point of RAM samples below zero, but if you want to add silence at the beginning of a RAM sample to create a delay, you can use the Insert Zero DSP function to add as much silence as you like. The start points of ROM samples can be set lower than zero (you can set negative numbers for the Start (S) and Alt (A) parameters). Doing this will cause portions of other samples to be played, which can create interesting effects.

Note that for RAM samples, you won't see any waveforms displayed to the left of the Start point, but you will for ROM samples.

The Alt (A) parameter lets you set a second, optional start or end point for the current sample. The Alt will be used when the Alt Switch parameter on the KEYMAP page is set to **On**, or when it's set to a specific control source and that control source is generating a value of more than +.5. (For example, if you assign **MWheel** as the control source for the Alt Switch parameter, the Alt will be used when the Mod Wheel—or whatever control source you have set to send MWheel—



is above its halfway point.) The Alt can be set before, after, or at the same point as the Start or End.

If you set the Alt after the End, you can extend the play of looped samples. Normally, looped samples will play through to the End, then will loop back to the Loop point, and continue looping like this until the note is released, when they go into their normal release. If the Alt is set after the End, looped samples will loop in the same way while notes are sustained. As soon as you release the notes, however, the samples will play through to the Alt point before going into release.

The Loop (L) parameter sets the beginning of the looped portion of the current sample. Although you can adjust this parameter while you're on the TRIM page, you'll normally want to adjust it while viewing the LOOP page, so you can see the loop transition points, enabling you to create as smooth a loop as possible. The Loop can be set at any point before the End, including before the Start and Alt. If you try to move it after the End, the End will move with it.

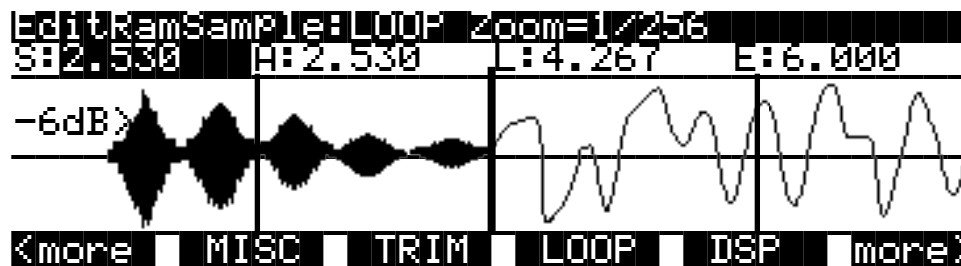
The End (E) parameter sets the point at which the current sample will stop playback. Typically you'll use this parameter to trim unwanted silence off the end of a sample, although you can use it to shorten a sample as much as you want.

If you want to truncate a sample to save memory, there are two points to keep in mind. First, if the Alt parameter is set before the Start, you won't save any memory by truncating the Start. Likewise, you won't save memory by truncating the End if the Alt is set after the End. You won't reclaim memory by truncating a sample until you save the sample and exit the Sample Editor.

You can also use the Truncate DSP function to automatically truncate your samples at a specified noise floor. As with the TRIM page, you'll recover memory after saving the sample and exiting the Sample Editor.

## The LOOP Page

The LOOP page features the same four parameters as the TRIM page, but the waveform display is quite different. The best way to understand what you see on the LOOP page is to switch back and forth between the TRIM and LOOP pages and study the waveform displays.



On the TRIM page you see the entire waveform—or as much of the waveform as your current zoom setting allows. When you move to the LOOP page, you'll notice that the page is split into two sections, left and right, divided by a vertical bar in the center. This bar is thicker than the vertical lines representing the Start, Alt, Loop, and End points, and does not move when you adjust any of these points.

To the left of the dividing bar you see the same segment of the current sample that you see on the TRIM page. The four vertical lines representing the Start (S), Alt (A), Loop (L), and End (E) points are visible. (Remember, you'll see all four vertical lines only if the values for the Start, Alt,

Loop, and End parameters are different.) To the right of the dividing bar you see the entire loop segment of the sample.

In the center of the loop segment is a dotted vertical bar that represents the loop transition point—that is, the point at which the sample reaches its End point and loops back to the Loop point. You can visualize the loop segment by starting at the vertical transition point; this is the beginning of the loop, as defined by the setting for the Loop parameter. The waveform progresses to the right, representing the initial portion of the loop segment. The waveform “disappears” off the far right side of the display, and “reappears” at the thick dividing bar at the center of the display. The waveform again progresses to the right, representing the final portion of the loop segment. It reaches the dotted vertical transition line, representing the End point of the sample, where it jumps once again to the loop point and repeats the cycle.

If you select the Loop (L) parameter and change its value, you’ll see the segment of the waveform to the right of the transition point shift its position. If you select the End parameter and change its value, you’ll see the segment of the waveform to the left of the transition point shift its position.

When you’re setting a loop segment for a sample, you’ll want to adjust both the Loop and End parameters so the two ends of the waveform meet (or come as close as possible) at the transition point. You’ll notice an audible click in your sample loop if the ends of the waveform do not meet at the transition point. The closer you can get the two ends of the waveform, the better the sound quality of your loop will be. With a bit of experimentation, you’ll develop the ability to create smooth loop transitions.

You’ll also want to try to set the loop point at a zero-crossing—a point where the sample waveform is neither positive or negative. Pressing the **Plus/Minus** buttons together will search (from left to right) for the sample’s next zero-crossing. You can usually press these buttons several times for any given sample without noticeably affecting the sound of the sample. If you press the **Minus** button, you’ll reverse the direction of the search, and the next time you press the **Plus/Minus** buttons together, the K2600 will search for the next zero-crossing to the left. Press the **Plus** button again to search toward the right.



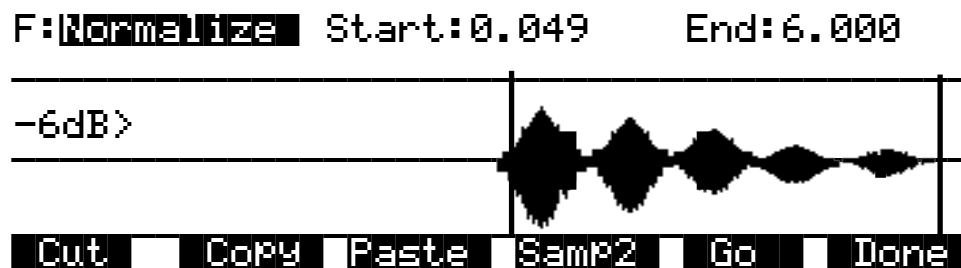
***Note:** The Gain affects the sensitivity of the zero-crossing search algorithm. The higher the gain, the more zero crossings it will find. At low gain settings, the algorithm will fail to find many or even all zero crossings.*

If you adjust the display Gain and Zoom of the sample while on the LOOP page, you’ll notice that the Gain affects the waveform on both sides of the loop point, while the Zoom affects only the left side of the page. You can’t zoom in on the loop transition point in the right half of the display.

You can also use the crossfade loop (XfadeLoop) DSP function to get a smooth transition between loop points. As with the TRIM page, you’ll recover memory after saving the sample and exiting the Sample Editor.

## The DSP Page (RAM samples only)

Select the DSP page with the **DSP** soft button. This gives you access to a long list of nonreal-time DSP functions, with which you can modify your RAM samples. The first time you select a DSP function, you'll see the Normalize function, shown below. Afterward, the most recently selected DSP function will appear when you select the DSP page.



All of the DSP functions operate on a segment of the current sample that you select before executing the function. In most cases, you'll use the Start and End parameters to define the start position and end position of the segment you want to modify. There are a few exceptions to this rule, which will be explained as applicable.

Please keep in mind that the Start and End parameters on the DSP pages are not the same as the sample Start (S) and End (E) parameters that you set on the TRIM and LOOP pages. When you're working on one of the sample DSP functions, Start and End position refer to the range of the sample that you want to process. Adjusting these parameters does not affect the overall start and end of the sample. It affects only the portion of the sample that you want to process. When you audition the sample by triggering a note, you'll hear only the range of the sample within the Start and End parameters on the current DSP page. To hear the entire sample, press the **Done** soft button to return to the Sample-editor page. Similarly, moving the S and E parameters on the TRIM or LOOP pages have no effect on the Start and End parameters on the DSP pages.

Use the **Plus/Minus** buttons or the Alpha Wheel to select the starting and ending positions of the selected sample segment (the Start and End parameters). You can audition the selected sample segment by triggering any note within the current key range.

The actual processing of the sample begins when you press the **Go** button. The K2600 will display a row of scrolling dots while it's processing. When it's finished, it will prompt you: Keep these changes? Trigger a note to hear the result. Press the **No** soft button to return to the page for the currently selected DSP function. This will undo any changes you've made. Press the **Yes** soft button to make the change. The Save dialog will appear (except for the Truncate function). You can save to a different ID than the one displayed, if you want to preserve the original sample. Pressing the **Cancel** soft button will return you to the current DSP page without saving the sample (the sample reverts to its original condition).

Six of the DSP functions (Mix, Mix Beat, Mix Echo, Insert, Replicate, and Beat Volume Adjust) involve selecting a second sample segment (Sample 2) to be processed with the currently selected sample. In these cases, use the **Samp2** soft button. When you press it, another page appears, enabling you to select a second RAM sample using the Alpha Wheel or the alphanumeric buttonpad. You can audition Sample 2 by triggering a note.

Once you've selected Sample 2, use the Start and End parameters to define the start and end positions of the segment of Sample 2 that you want to process. While you're on the Samp2 page, you'll hear the currently selected Samp2 when you trigger a note. When these positions are

defined, press the **OK** soft button to return to the DSP page where you can continue your editing.

Once you've selected a sample on the Samp2 page, it remains selected until you return to the Samp2 page and select another sample. You can always use the Samp2 page to audition a second sample.

You can use the **Copy** soft button on the Samp2 page to copy the selected segment to a buffer, then paste the segment into Sample 1 when you return to the DSP page.

## The Soft Buttons on the DSP Page

### Cut

This button will cut (remove) the currently selected sample segment from the currently selected sample, and store it in a buffer. This is equivalent to cutting a section out of a piece of audio tape and splicing the remaining ends together. The cut segment is then available for pasting elsewhere in the sample. The segment you cut will remain in memory until you replace it with another cut or copy command, or until the K2600 is shut off. If you accidentally cut a sample segment, you can restore it by immediately pressing the **Paste** soft button.

Compare this function to the Delete DSP function, which removes the selected sample segment, but doesn't store it in a buffer. If you delete a sample segment, it's *gone*.

### Copy

Use this button to store the selected sample segment in a buffer without altering the current sample. The copied segment will remain in memory until you replace it with another cut or copy command, or until the K2600 is shut off.

### Paste

This button has an effect only after you've cut or copied a sample segment using the Cut or Copy soft buttons. The **Paste** soft button inserts the contents of the Cut/Copy buffer after the start position of the currently selected sample segment. This is like splicing a section of audio tape into another section; it extends the length of the sample. You can undo a paste by pressing the **Cut** soft button immediately after pasting.

When you cut or copy a sample, it's stored in a buffer. Whatever is stored in the buffer will remain available for repeated paste commands until replaced by another cut or copy command, or until the K2600 is shut off or reset (soft or hard). The buffer is also cleared if you execute a master delete, and select "Delete everything."



*Note: Cutting and pasting always "save" the resulting sample, so you can't use the exit-without-saving short cut to undo a cut or paste operation.*

### Samp2

The **Samp2** button, as described earlier in the DSP page section, enables you to select a segment of a second RAM sample to be processed with the currently selected sample.

### Go

Press the **Go** soft button when you want to execute the currently selected DSP function on the currently selected sample. When processing is completed, the K2600 will prompt you to keep

the changes. You can audition your changes and decide whether to keep them, then press **Yes** or **No**.

### Done

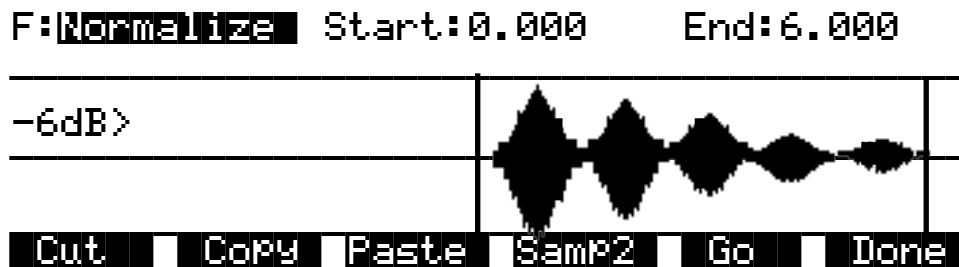
Press the **Done** soft button to return to the previously selected Sample-editor page when you're finished with the DSP functions.

## DSP Functions

Once you've entered the Sample Editor, press the **DSP** soft button to gain access to the DSP functions. The DSP function parameter will be highlighted, allowing you to scroll through the list of functions with the Alpha Wheel or **Plus/Minus** buttons.

Remember that the DSP functions operate only on RAM samples. You'll notice that if you're editing a ROM sample, there is no **DSP** soft button; instead, there's an extra **Link** soft button for your convenience.

### 1 Normalize



With the Normalize function, you can rescale the amplitude of the selected sample segment to optimize its level relative to other samples. The Normalize function will uniformly boost the amplitude of the current sample range as high as possible without clipping, stopping just before the loudest element of the sample would be clipped. You might want to use the Volume Adjust function to boost the current segment manually, but the Normalize function does it automatically, and prevents you from boosting the amplitude too much.

### 2 Truncate

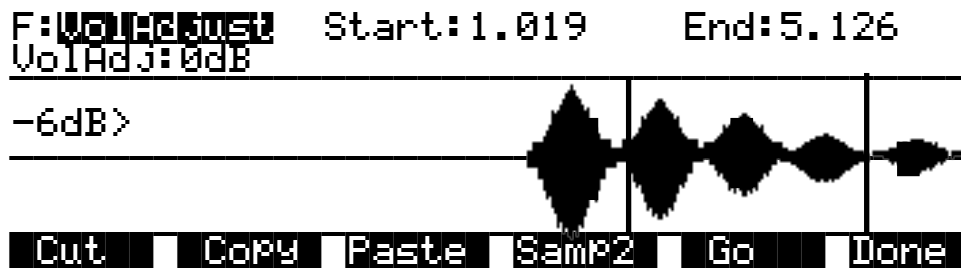


The Truncate function will automatically reset the Start (S) and End (E) points of the sample. This can be quicker than trimming the sample manually on the TRIM page.

Use the Start and End parameters to select a specific sample range. Set the Thresh parameter from **-96** to **0dB** to set the noise floor. When you press the **Go** soft button, the K2600 will search inward from the start and end points you set, until it finds the first sample that exceeds the noise floor. Everything outside the range will be left out. You'll want to experiment with different thresholds to find the noise floor that suits each sample you truncate. If the new End point is inside the current loop point, the loop will be disabled.

When you press the **Go** soft button, you'll be prompted "Keep this change?" Unlike the other DSP functions, answering **Yes** does not bring up the Save dialog. It registers the new Start and End positions as the Start (S) and End (E) points (as set on the TRIM or LOOP page). In fact, you can go to either of these pages to readjust the S and E points if you want to include more of the sample. To save the new S and E points, press the **Done** soft button (if necessary) to return to the SampleMode page, then press either the **Save** soft button, or **Exit**.

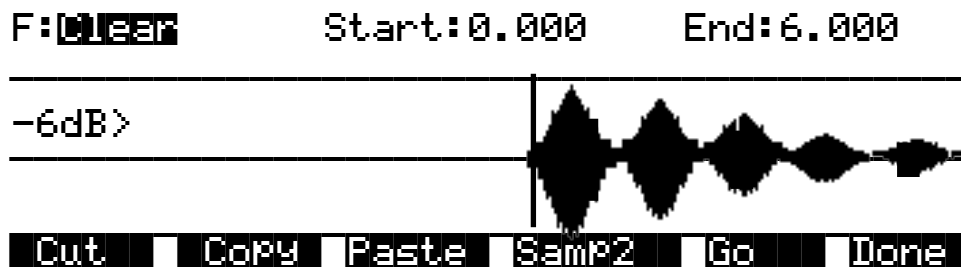
### 3 Volume Adjust



Use this function for a uniform cut or boost in the amplitude of the selected sample segment. This function will clip samples if you adjust the volume too high. This won't hurt the K2600, and you may find it useful in some applications. In any case, you'll need to choose your start and end points carefully, if you want to avoid abrupt changes in volume.

When you've selected the range to be adjusted, select the VolAdjust parameter with the cursor buttons, and use the Alpha Wheel to adjust the volume of the selected range. You can cut/boost the volume from -96 to 96 dB.

### 4 Clear



The result of this function is like erasing a section of recording tape. Use it to create sections of silence without changing the overall length of the sample. If you want to completely remove a segment and shorten the sample, you can do it with the Delete function.

### 5 Delete



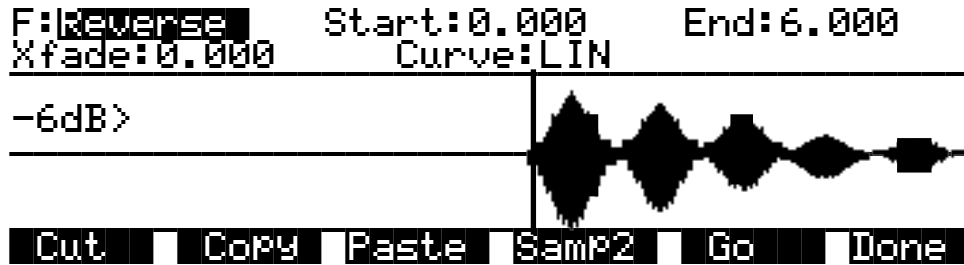
Unlike the Clear function, this will erase the samples within the selected range and shorten the sample, like cutting a section out of a tape and splicing the ends. If you want to silence a segment of the sample without shortening it, use the Clear function.

The Crossfade (Xfade) parameter enables you to smooth the transition from the deleted sample segment to the remaining sample segments, and can create overdub effects. The value of the Xfade parameter defines the amount of time the sample will take to fade to silence at the beginning of the deleted segment, and to ramp back up to the volume of the remaining portion of the sample.

The Curve parameter lets you choose from a variety of crossfade curves that affect the nature of the crossfade. A value of **LIN** gives a straight linear curve that fades uniformly from start to finish. A value of **EXP** sets an exponential curve that starts gradually and steepens toward the finish. A value of **COS** sets a Cosine curve. A value of **EQL** applies an equal crossfade that is the same at both ends of the deleted segment. A value of **MIX** assigns a curve that will start the crossfade before the start point of the deleted segment, and will end after the end point of the deleted segment. This differs from the other curves, which apply crossfade only to the deleted segment. See *Crossfade and Volume Adjust Curves* on page 14-34 for a diagram that helps to explain these curves.

Crossfading a sample will shorten it by half the time you specify. The greatest length of crossfade you can get is the entire length of the shorter crossfaded segment (even though you can enter greater amounts with the Alpha Wheel or alphanumeric buttonpad). A crossfade of this length would crossfade the two samples over the entire length of the shorter segment.

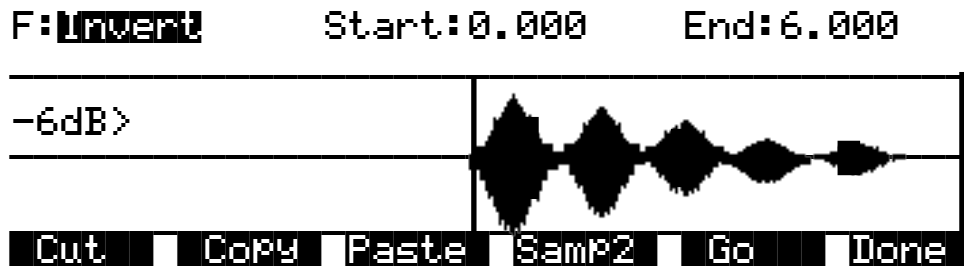
6 Reverse



With this function you can reverse the order of the individual samples between the start and end positions you set. The Xfade parameter lets you apply a crossfade to the start and end of the reversed segment. The Curve parameter lets you select a crossfade curve. The available values are LIN, EXP, COS, EQL, and MIX. These curves are described on page 14-34.

Like the crossfade parameter in the Delete function, this crossfade will also shorten the sample. The maximum crossfade length is half the length of the reversed segment.

7 Invert



Use this function to invert the waveform of the selected sample range. This reverses the phase of the sample. You won't hear any difference unless it is played in reference with the original sample.

The results will vary depending on the type of sample. For an interesting phase effect, make a copy of a sample, then invert the copy, then assign it to the keymap of a second layer in a program that uses the original sample in the keymap of the first layer (or set them as the Left and Right samples in a stereo keymap). Add a delay to one of the layers or keymaps to create the phase effect.



## 8 Insert Zero



This function will insert a period of silence of any length into the selected sample range. This function is equivalent to splicing a section of blank tape into an existing segment of recorded tape. It's useful for creating a delay on the start of a sample, or creating a silent spot in the middle or at the end of the sample. Adjust the Start parameter to determine the point at which the period of silence will begin. Adjust the Length parameter to specify how long the silence will be. The End parameter has no effect for this function.

## 9 Mix



With this function you select a segment from Sample 2, and merge it with the selected segment from Sample 1, beginning at the point you set with the Start parameter. This is equivalent to mixing two audio signals through a mixing board. If the Sample 2 segment is longer than the segment from the Start (S) to the End (E) of Sample 1, the resulting sample will be longer than the original Sample 1.

Use the In and Out parameters to specify the length of time it takes Sample 2 to reach full amplitude and to fade to silence. The Curve parameter selects the curve of the fades. The available values are LIN, EXP, COS, EQL, and MIX. These curves are described on page 14-34.

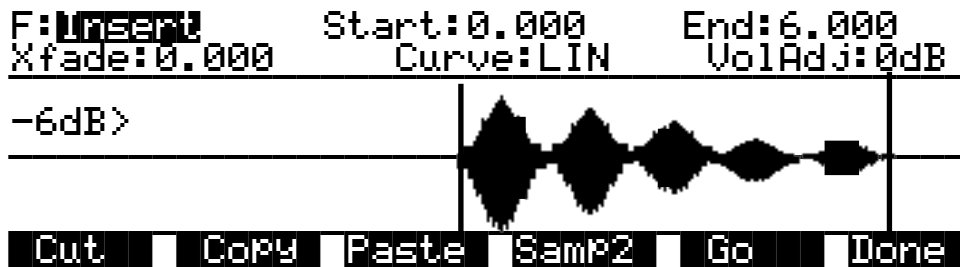
The Volume Adjust parameter will cut or boost the amplitude of Sample 2 from -96 to 96 dB before merging.

If the sample rate of Sample 2 is different from that of Sample 1, the K2600 will alert you that the sample rates differ. If you mix samples with different rates, you'll hear a pitch shift in the mixed sample. If you don't want this pitch shift, use the Resample function to match the sample rates of the two samples before mixing.

**Mixing Samples, Step-by-step**

First, use the Start parameter of Sample 1 to set the point at which the mix will begin. Then press the **Samp2** soft button, and a page will appear enabling you to view and audition the list of RAM samples. Select a sample with the Sample parameter, then use the cursor buttons to select the Start parameter. Turn the Alpha Wheel to set the beginning of the segment to be mixed in. Repeat this process for the end position of the sample. You can audition the sample as you're setting the range. Once the start and end points are set, press **OK** to return to the DSP page. Then press **Go** to initiate the mix.

**10 Insert**



Use this function to insert the selected segment from Sample 2 into Sample 1. This is like splicing a section of tape into an existing tape. This differs from the Mix function, which merges the two samples into one.

Use the Crossfade parameter to control the crossfades at the start and end of the inserted sample. The Curve parameter selects the curve of the crossfade. The available values are **LIN**, **EXP**, **COS**, **EQL**, and **MIX**. These curves are described on page 14-34.

The Volume Adjust parameter will cut or boost the amplitude of Sample 2 from -96 to 96 dB before inserting.

If the sample rate of Sample 2 is different from that of Sample 1, the K2600 will alert you that the sample rates differ. If you insert a sample with a different rate, you'll hear a pitch shift in the inserted sample. If you don't want this pitch shift, use the Resample function to match the sample rates of the two samples before inserting.

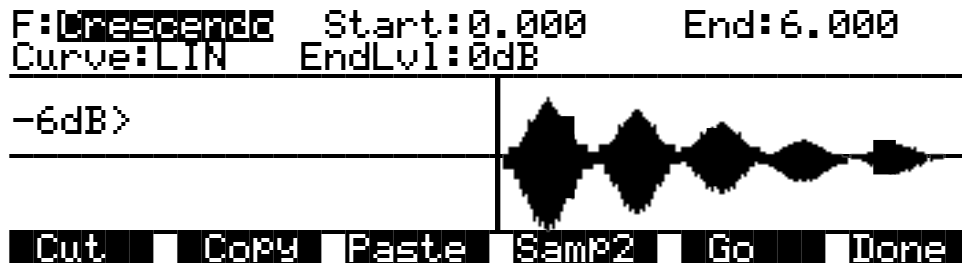
### 11 Volume Ramp



This function lets you apply a ramp to the volume of the selected sample range. The Start Level and End Level parameters let you set the amount of cut (negative value) or boost (positive value) at the start and end points of the segment. The Curve parameter determines the shape of the ramp that scales the amplitude of the sample between the start and end amplitudes. The available values are **LIN**, **EXP**, **COS**, **EQL**, and **MIX**. These curves are described on page 14-34.

The VolRamp function affects only those samples within the start and end points. The sample will clip if you apply large amounts of volume ramp. If you want to ramp the volume of a sample segment up or down, then keep the volume at that level, use the Crescendo function.

### 12 Crescendo/Decrescendo (Crescendo)



Similar to Volume Ramp, this function applies a curve that scales the amplitude of the selected sample segment. Unlike Volume Ramp, however, you simply select a start and end point, and a single level. The amount of cut or boost starts at 0dB at the start point of the ramp, and reaches the level you specify when it reaches the end point of the ramp. The samples after the end point are scaled to the amplitude level of the end point.

When you set a negative end level (decrescendo), the sample's volume is cut by the amount you specify. If you set a positive end level (crescendo), the sample segment is first cut by the amount you specify, then is boosted back to its original level. This enables you to add large crescendos without clipping the sample. You may want to adjust the gain of the layer using the sample to match its level with other sounds.

Use the Curve parameter to select the shape of the curve within the selected segment. The available values are **LIN**, **EXP**, **COS**, **EQL**, and **MIX**. These curves are described on page 14-34.

### 13 Resample



Use this function to change the sample rate of the samples in the selected segment. This is convenient for converting samples to new rates matching those of other samples to be mixed with or inserted into. It's also useful for saving memory and altering the timbre of a sample.

If you include the entire sample in the segment, the new rate will be applied to the entire sample, and will be saved with the sample. If you select a shorter segment, only that segment will be modified, and it will sound pitch shifted relative to the remainder of the sample. To resample so that you'll hear a higher pitch for the selected segment, select a lower sample rate. For a lower pitch at the selected segment, choose a higher sample rate. This is because the K2600 applies the same playback rate to the entire sample, and doesn't compensate for the differing sample rates of the sample segments.

If the loop points of a looped sample are included in the segment to be converted, the K2600 will ask you if you want to adjust the rate slightly to optimize the loop. Press the **No** soft button if you don't want the rate adjusted.

You can use the Quick parameter to select from two resampling routines. Use **Quick 0** to get an idea of the sound, then use **Quick 1** for your final take.

### 14 Time Warp



With this function you can change the length of the selected sample segment without affecting the pitch. This function applies sophisticated routines that lengthen or shorten the selected sample segment to play it back over a different time period, modifying the playback rate so the pitch remains unchanged.

The Start and End parameters define the segment to be processed. Use the New Length (NewLen) parameter to specify how long you want the resulting sample to be. While the

function is in progress, the display will indicate the percentage of individual sample segments that have been processed.

The Quick parameter lets you select one of three warping routines. Use **Quick 0** to audition your new sample, then use **Quick 1** or **2** for your final take. **Quick 2** takes more time to process, but gives you a better final result.

This function is extremely useful for fine adjustments in the length of a sample. You can also apply greater amounts of warp for a wide range of effects. Experimentation will give you an idea of the amount of alteration you want to apply.

### 15 Pitch Shift



The PitchShift function is the counterpart of the TimeWarp function; it shifts the pitch of the selected sample segment without changing the playback time—very useful for tuning samples when the playback time is crucial. Use the Start and End parameters to define the segment to be shifted. The Shift parameter determines the amount of pitch shifting, up to  $\pm 30000$  cents.

Like the crossfade parameter in the Delete function, this crossfade will also shorten the sample. The maximum crossfade length is half the length of the reversed segment.

The Quick parameter lets you select one of three shift routines. Use **Quick 0** to audition your sample, then use **Quick 1** or **2** for the final take. **Quick 2** takes longer to process, but gives you better results.

### 16 Mix Beat



With this function you can mix the selected range of Sample 2 into Sample 1 at regular intervals. One natural application of Mix Beat is to mix various percussion samples (Sample2) on different beats of Sample 1. You could then loop Sample 1, creating a drum-loop sample out of individual percussion samples. You don't have to use percussion samples, however. You can create many

interesting and unusual effects by mixing different types of samples, especially if you set the Tempo parameter to a high value.

The Tempo parameter sets the interval between repetitions in beats per minute. The Of parameter establishes the number of beats per measure, and the Beat parameter sets which beat gets mixed. For example, if you set the Tempo parameter to 120, you'll have two beats per second. If you set the Beat parameter to 1, and the Of parameter to 1, Sample 2 will be mixed in twice per second, for the duration of the selected range of Sample 1. The first mix of Sample 2 occurs at the Start (S) point of Sample 1. If you change the Of parameter to 4, Sample 2 will be mixed into Sample 1 on the downbeat of every measure of 4.

The length of the ranges you set for Sample 1 and Sample 2 affects the results of the mix. In the above example, if Sample 1 is two seconds in length, and the mix of Sample 2 is on Beat 1 of 1, you'll mix four segments of Sample 2, at half-second intervals. On beat 1 of 4, you'd hear just one mix of Sample 2, right at the top.

To set up a MixBeat, first select the desired segment of Sample 1 using the Start and End parameters. Then press the **Samp2** soft button to select the sample to be mixed in, and the selected range of that sample. Note that when you go to the Samp2 page from the Mix Beat page (and from the Replicate and Mix Echo pages), an extra parameter—Increment (Incr)—appears. This parameter isn't available for the other DSP functions.

You can use the Incr parameter to shift the starting position of Sample 2. If you set a positive value, the start and end points of Sample 2 will shift later each time it's mixed. A negative value will shift the start and end points of Sample 2 earlier with each mix.

To optimize processing time, keep the range of Sample 2 shorter than the interval between mixed-in segments. In the example above, if the mix were on beat 1 of 1, you'd want to keep Sample 2 at a range of a half second or less. If it were on beat 1 of 4, the range of Sample 2 could be as much as two seconds.

Next, press the **OK** soft button to return to the Mix Beat page, then use the Tempo parameter to select the rate at which Sample 2 will be mixed in. You can choose a tempo from 1 to 9999 beats per minute. Try setting the tempo at 9000 or more, and mix a very small range of Sample 2 into Sample 1 to create a wide variety of periodic waveforms.

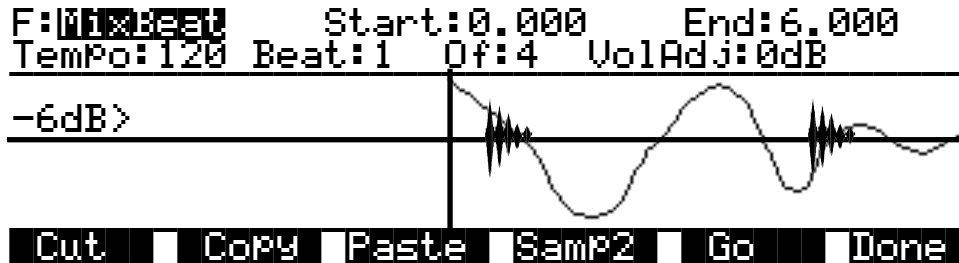
Now set the values of the Beat and Of parameters to determine how the mixed sample will repeat. The Beat parameter determines the beat (s) on which Sample 2 will be mixed—from 1 to 9999. The Of parameter determines the measure length—also from 1 to 9999.

Finally, use the VolAdjust parameter to set the volume of the mixed sample segment—from -96 to 96 dB.

If the Of parameter is set to a value of 0, the Sample 2 segment will be mixed in on every beat, regardless of the setting for the Beat parameter. If the Beat parameter is set to a negative value, the segment of Sample 2 that's mixed in will be moved forward in time by the length of one beat each time it's mixed in; that is, you'll hear a later portion of the sample. Another way to accomplish this is to use the Increment (Incr) parameter on the Samp2 page. Set it to a positive value to use a later portion of Sample 2 with each repetition, or a negative value to use an earlier portion. The Beat parameter must be set to a value of 0 or higher for this to work.

For example, suppose you've chosen a six-second sample as Sample 1, and you use the entire sample as the selected segment. You also select a half-second segment of Sample 2 to be mixed in. If you choose a Tempo value of 120 beats per minute (2 beats per second), there will be 12 beats within your six-second Sample 1 segment. If you set the Beat parameter to 1 and the Of

parameter to 4, then Sample 2 will play on the first, fifth, and ninth beat of your six-second Sample 1 segment. The result will look something like the diagram below.



If you change Beat to 3, Sample 2 will play on beats 3, 7, and 11. If you set Beat to 1 and Of to 6, Sample 2 will play on beats 1 and 7. If you set Beat to -1, Sample 2 will still play on beats 1 and 7, but the sound of the sample will change each time, as the start point of the Sample 2 segment moves forward (later) by one beat each time it is mixed in.

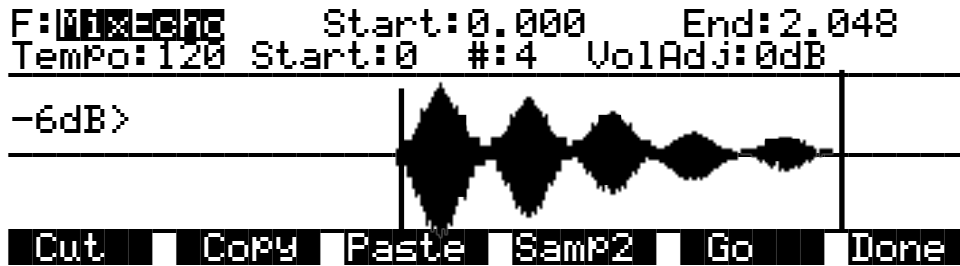
Mix Beat (as well as Replicate and Mix Echo) may seem a bit complicated at first, but if you experiment with different settings—especially for Beat and Of—you'll quickly begin to get a feel for what these functions do.

### 17 Replicate



With the exception of two differences, the replicate function is similar to the Mix Beat function. The primary difference is that the replicate function uses the selected Sample 2 segment to overwrite (replace) the selected Sample 1 segment, instead of merging the two segments like the Mix Beat function. This makes it run faster. The other difference is that there is a crossfade parameter instead of a Volume Adjust parameter. The crossfade parameter lets you smooth the transition points from Sample 1 to Sample 2.

18 Mix Echo



This function operates much like Mix Beat, but instead of the Beat and Of parameters, you have Start and # parameters. The Start parameter sets the beat at which the selected Sample 2 segment begins being mixed with the selected Sample 1 segment. Sample 2 is repeated on every beat after this starting point. The # parameter determines how many times the Sample 2 segment is repeated.

The Volume Adjust parameter will affect the volume of each repetition of Sample 2. It sets the relative level of the last beat, and boosts or cuts each repetition by a proportionate amount from the starting level. This is a linear adjustment, measured in decibels, evenly spaced over the total number of mixed-in beats.

If you set the # parameter to a positive value, the adjustment in volume begins on the second mixed-in beat (the first beat plays at the initial level). If you set a negative value, the first mix of Sample 2 will be cut or boosted by the Vol adjust amount.

19 Beat Volume Adjust

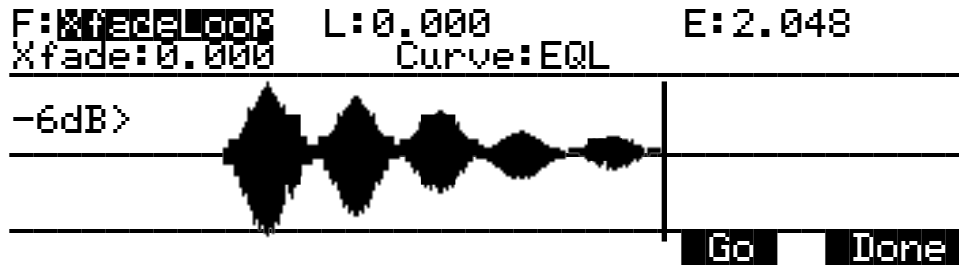


This function works much like Mix Beat. The difference is that you're not mixing in any samples, you're just volume-adjusting Sample 1 at regular intervals. Each beat is adjusted in volume by the amount specified for the VolAdj parameter. This is useful for mixing part of a sample within itself.

Set the Start and End positions of Sample 2 to define the length of time over which the volume will be adjusted. For example, if you had a six-second sample, and you set the Tempo to 120, the Beat parameter to 1, and the Of parameter to 1, you'd have twelve beats defined. If you then set the range of Sample 2 at a half-second, and set the VolAdj parameter to -3 dB, then you'll insert 12 volume adjustments into Sample 1. Each adjustment lasts a half second, and brings the level down 3 dB.



## 20 Crossfade Loop (XfadeLoop)



The Crossfade Loop function lets you create smoother loops by crossfading the beginning segment of the loop with a segment of equal length at the end of the loop. These segments can be defined by the Loop and End parameters as set on the TRIM or LOOP page for the current sample, or with the Loop and End points on the XfadeLoop page. Changing the Loop and End parameters on the XfadeLoop page will change them on the TRIM and LOOP pages, and vice versa. Using this function is equivalent to setting the loop on the LOOP page, but with the added feature of a crossfade at the loop transition point.

The Xfade parameter determines the length of the crossfade, while the Curve parameter sets the shape of the crossfade curve. The available values are LIN, EXP, COS, EQL, and MIX. These curves are described on page 14-34.

## 21 Dynamics



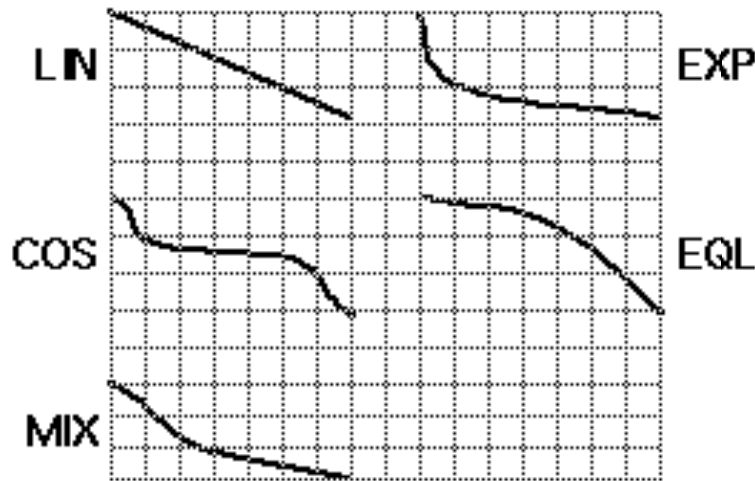
The Dynamics function is a general-purpose compressor with a few features that make it suitable for creating smooth sample loops. First, while most compressors set a threshold relative to 0 dB, the Dynamics function sets its threshold relative to the peak of the signal you're compressing. Secondly, after compressing the signal, it adjusts the volume of the compressed segments (there may be more than one) so that they match the uncompressed segments as closely as possible.

Set the Start and End to select a segment of the signal for compression. The Thresh parameter sets a threshold in dB, relative to the peak amplitude of the selected segment. Any portion of the segment between Start and End that exceeds that threshold gets compressed. The Comp parameter determines how much compression gets applied to sample points that exceed the threshold, expressed as a percentage of the amount by which the sample points exceed the threshold. The Time parameter controls the length of time that the compressor ramps back down to zero compression when it reaches the end of a compressed segment. This helps assure a smooth transition between compressed and uncompressed segments.

Here's an example. Suppose you set Thresh, Comp, and Time as shown in the preceding diagram. Beginning from the Start point, when the signal first reaches the threshold, the Dynamics function begins compressing the signal, attenuating every point above the threshold by 50%. Compression continues until the signal goes back below the threshold. At that point, the compression starts diminishing, and ramps back to zero in 0.3 seconds. If the signal goes above the threshold again within the selected segment, this process repeats.

## Crossfade and Volume Adjust Curves

There are five curves that can be applied to a number of DSP functions: LIN, EXP, COS, EQL, and MIX. The LIN curve is a straight linear curve, which will create an even cut or boost. The EXP curve is exponential, that is, gradual at one end and steep at the other. The COS curve is a segment of a cosine curve, which is relatively steep at both ends and flatter in the middle. The EQL curve steepens at an even rate, approximating an equal-power fade curve. The MIX curve is a gradual curve that approximates manually dropping or raising the faders on a mix board. The diagram below shows each of the curves as it would be applied to a cut in amplitude.



## The Keymap Editor

The Keymap Editor lets you customize the K2600's factory preset keymaps and save them to RAM. You can also build your own keymaps from scratch.

Keymaps are an integral part of every layer of a program. Each keymap contains a set of parameters determining which sample(s) the K2600 will play when you trigger a note. Each layer has at least one keymap, but it can have two keymaps when you're working with stereo samples. Each of these stereo keymaps uses two of the 48 available voices.

Each keymap consists of a set of key (note) ranges—C 4 to G 4, for example. The entire span of each keymap is from C 0 to G 10. Each range has a sample root assigned within the range. Each sample root is a distinct ROM or RAM sample. Within each key range, the sample root is transposed up and down to play on each of the range's notes. You can view each range by changing the value of the Key Range parameter on the Keymap-editor page. You can mix samples of different timbres within a single keymap, and even tune individual keys to any pitch by defining key ranges to single notes and assigning samples to each of those notes.

When you trigger a note, the K2600 identifies the key range where the Note On event occurred. It also checks the attack velocity value of the note. It then addresses its memory, and retrieves the sample root that's assigned to that key range and attack velocity value. If the note that's triggered is not the note where the sample root is assigned, the sample is transposed to play at the correct pitch. The K2600 then generates the digital signal that represents the sound of the note. At this point the keymap's job is done, and the signal proceeds through the layer's algorithm and on to the audio outputs.

You can assign as many key ranges to a keymap as you like, even creating a separate range for each note. This would allow you to tune each key independently, to create microtonal tunings. For keymaps that use a single timbre, like the Grand Piano, there's a key range for each sample root stored in memory. For acoustic instrumental sounds, the more key ranges you have for a keymap, the more realistic the sound will be, since there will be less pitch shifting of the sample root within the key range.

Of course, you can assign sample roots with different timbres within the same keymap. Many of the drum kit keymaps in ROM, for example, have about 20 key ranges, with several different timbres assigned as the sample roots. You can also create a keymap with a single key range that spans from C 0 to G 10, if you want to stretch a single sample root from C 0 to G 10. Keep in mind, however, that samples can be transposed upward only to a limit of 96 KHz for the playback rate. For 48K samples, that's an octave of upward transposition—or two octaves if you set the SmpSkp parameter (on the KEYMAP page in the Program Editor) to **On** or **Auto**. Samples can be transposed downward without limit.

Think of a keymap as if it were a single piece of string, divided into different sections that adjoin one another. Sections cannot overlap. If you have one range that goes from C4 to F4 and another that goes from F#4 to C5, then if you change the first range to be C4 to G4, the second one will change to be G#4 to C5.

Also, you can't have "nothing" assigned to a key range. Even if it is Silence (#168), there will always be a sample assigned to every range in the keymap. This is something to watch out for when creating drum programs. For example, let's say you are creating a program with 20 layers. Each layer has its own keymap, which has just one sample assigned to part of the keyboard with the rest of the key range assigned to Silence. Make sure that you limit the note range of each layer using the LoKey and HiKey parameters on the LAYER page in the Program Editor. If each layer covers the entire range, then each note you played would trigger 20 voices (one for each layer). You would only hear one drum per note because all the other layers are triggering "Silence." Because of the voice-stealing algorithms in the K2600, the voices would almost immediately become available again, since they have no amplitude. But for one brief instant, the voice would be triggered, which could cause other voices to be cut off.

You can also create multi-velocity keymaps—that is, keymaps that will play different timbres depending on the attack velocities of your Note On events. Program **12 DynEPiano^EPnoPF**, for example, uses a keymap with two velocity ranges. Each key range in a multi-velocity keymap contains two or more distinct sample roots that the K2600 chooses between, according to the attack velocity of the note. When in the Keymap Editor, you can select the different velocity ranges within a keymap using the **Chan/Bank** buttons. To create your own multi-velocity keymaps, Select any program (like 199 Default Program), and press **Edit** to enter the Program Editor. Then select the KEYMAP page, and select one of the multi-velocity keymaps as the value for the Keymap parameter (the multi-velocity keymaps are IDs **31–38**). The description of the Velocity Crossover parameter on page 14-39 has more information.

The Keymap Editor is nested within the Program Editor. The first step in using the Keymap Editor is to select the keymap you want to edit. This is done on the KEYMAP page in the Program Editor, using the Keymap parameter. Once you've done this, just press the **Edit** button, and you'll enter the Keymap Editor. If you want to edit a different keymap, return to the KEYMAP page in the Program Editor and select the desired keymap. If you want to build a keymap from scratch, start with the keymap **168 Silence** (see *Building a Keymap* on page 14-39).

This keymap template contains one key range from C 0 to G 10, and is a convenient starting point for adding key ranges and assigning sample roots. The Keymap-editor page looks like this:

```
editKeymap <>VelocityRange:PPP-fff
MasterXpose :0ST
Key Range   :C 0-G 10 Lo:C 0 Hi:A 1
Sample      :168 Silence-C 4
Coarse Tune :0ST
Fine Tune   :0ct
VolumeAdjust:0.0dB VelCrossover:None
Name Save Delete Dump NewRng Assign
```

Parameter	Range of Values
Master Transpose	-128 ST to 127 ST
Key Range	Variable
Low Key	C 0 to G 10
High Key	C 0 to G 10
Sample	Sample Root list
Coarse Tune	-120 to 60 semitones
Fine Tune	-49 to 50 cents
Volume Adjust	± 48 dB
Velocity Crossover	None, ppp to fff

The top line of this page tells you which velocity range you’re currently looking at. If the current keymap is a multi-velocity keymap, the **Chan/Bank** buttons let you select between the velocity ranges. The velocity range is set with the Velocity Crossover parameter(s), described on page 14-39.

## The Soft Buttons in the Keymap Editor

The first four soft buttons execute the basic library functions, enabling you to name, save, or delete the current keymap, or dump it via a MIDI SysEx message.

### New Range (NewRng)

The **NewRng** button lets you define a range to edit, whether it’s to assign a different sample, or to adjust the pitch or volume. Just press **NewRng**, then trigger the note you want as the low note, then the high note. The K2600 will prompt you for each note. When you trigger the high note, you’ll return to the Keymap-editor page, and the edit range you defined will be selected. The next change you make will affect only that edit range.

There’s more than one way to use this function. If you set an edit range that’s completely within an existing key range, you can modify the edit range without affecting the rest of the key range or the adjacent key ranges. If you set an edit range that overlaps part or all of another key range, the sample assigned to the lower key range will be applied to the entire edit range. This is an easy way to define a new key range that replaces one or more existing key ranges.

## Assign

The **Assign** soft button lets you select a sample, then specify the key range to which it's assigned. This enables you to insert a new key range within the current keymap. When you press the **Assign** soft button, a dialog appears that prompts you to select a keymap from the Keymap list. Scroll through the list, then press the **OK** soft button. You'll then be prompted to define the new key range by triggering the notes you want to be the lowest and highest notes of the range. (Press the **Cancel** soft button if you change your mind.) When you trigger the low and high notes, the new key range is inserted. If the new key range partially overlaps an adjacent key range, the existing key range will be adjusted to accommodate the new range. If the new key range completely overlaps an existing key range, the original key range will be replaced.

## Special Double Button Presses in the Keymap Editor

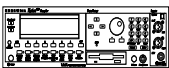
Suppose you have a sample whose root key is C 4, and you want to assign it to A 0, because you don't expect to play it often. If you want it to play back without transposition, you'll have to adjust the Coarse Tune parameter. Calculating the right value for Coarse Tune can get tedious if you're assigning a large number of samples. Fortunately, there's a short cut.

1. Assign a sample root to a key range, either using the Lo, Hi, and Sample parameters or using the **Assign** soft button.
2. Highlight the value of the Coarse Tune parameter.
3. Press the **Plus/Minus** buttons at the same time. The value of Coarse Tune changes automatically. If the sample is assigned to one note, the K2600 sets Coarse Tune so that the note plays the sample without transposition. If the sample is assigned to a range of notes, the K2600 sets Coarse Tune so that the middle note of the range plays the sample without transposition.

## Keymap Editor Parameters

### Master Transpose

This parameter does not really pertain to the keymap itself. Instead it is identical to the Transpose parameter found on the MIDI-mode TRANSMIT page. If you change the value here, the same value will be reflected on the MIDI-mode TRANSMIT page, and vice versa. It transposes the entire instrument globally. The reason it is placed on this page is that it will allow you to assign samples across the entire keyboard easily, when you are using a keyboard that has fewer than 88 notes.



As with the Transpose parameter on the MIDI-mode TRANSMIT page, if you have the rack-model K2600, this parameter will not transpose without using the Local Keyboard Channel parameter (refer to page 10-7 for more on the Local Keyboard Channel).

### Key Range

This parameter shows you which key range you're currently viewing or editing. Changing the value of this parameter selects a different key range, and shows you the sample assignment of that key range, as well as other parameters related to that key range. Use this parameter to move from one key range to another within the keymap, and assign different samples without adding new key ranges, as well as editing the transposition, tuning, volume and velocity crossover (if any) of the sample root assigned to the key range.

### Low Key (Lo), High Key (Hi)

With these parameters you can use any of the data entry methods to change the low and high notes of the current range. These parameters let you extend or shorten the width of a key range. You can extend a key range to the full capacity of the K2600 (C 0 to G 10).

The setting for the low key cannot be higher than the setting for the high key. Similarly, the setting for the high key cannot be lower than the setting for the low key.

### Sample

This is where you assign a sample root to the current key range. Depending on the nature of the sample root—an individual sample or a block of sample roots—the sample's name looks a bit different in the display. Each sample's name consists of three parts: a numeral, a name, and a note number—for example, **1 Grand Piano G#1**.

The numeral is the sample block ID. If the sample object is an individual sample, the sample block ID is the same as the sample's object ID. If the sample object is a group of sample roots, the object ID of the first root in the group determines the sample block ID. The remaining roots in the block have the same ID, and differ only in their note numbers.

Next comes the name of the sample, which typically describes the sample's timbre. The final part of the sample's name refers to the pitch at which it was originally sampled. For many timbres, multiple samples are made at various pitches. As you scroll through the Sample list, you'll see only the pitch of the sample change until you reach the next sample block.

Highlight the Sample parameter, hold the **Enter** button, then play a note to see the complete name of the sample played by that note. For example, when you play middle C on a piano program you'll see a name such as **1 Grand Piano C4**. Move down the keyboard an octave and the sample will be **1 Grand Piano G#2**.

### Coarse Tune

Coarse Tune allows you to transpose a sample for a given range. This is extremely useful when you have set the Root key of the sample for one note but want to assign the sample to a different part of the keyboard and still be able to play it without transposition. For example, if you originally set the Root key at C4 but want the sample assigned to C3, you would set Coarse Tune to 12ST, transposing it up one octave. Now the original pitch will play at C3, one octave down. If you examine the drum and percussion kit keymaps in ROM, you will see that we have done this. Most of our ROM drum samples have the Root key set at C4.

There's a short cut for adjusting the Coarse Tune automatically so that the sample plays with minimal transposition in the assigned key range. See *Special Double Button Presses in the Keymap Editor* on page 14-37.

### Fine Tune

This gives you further pitch control. Once the sample's pitch is close to the desired level, use the Fine tune to sharpen or flatten it as much as a half-semitone.

### Volume Adjust

Here you can adjust the volume of the notes in the current key range. This enables you to make each key range play at the same volume even if the samples in the various ranges were recorded at different volumes.

### Velocity Crossover (VelCrossover)

This parameter applies only when the keymap assigned to the currently selected program is a multi-velocity keymap. The name of the keymap usually indicates whether it's of the multi-velocity variety (**29 Bass^Slap Bass**, for example). Multi-velocity keymaps have a predetermined number of velocity levels, each of which can be assigned a different sample.

The K2600 supports keymaps with up to eight velocity levels. You can't add velocity levels to existing keymaps; if you want to create your own multi-velocity keymaps, select an existing multi-velocity keymap in the Program Editor before entering the Keymap Editor. Then you can select the different velocity levels with the **Chan/Bank** buttons, and assign samples to the different levels. The currently selected velocity range is shown in the top line of the display.

When the current keymap is a single-velocity keymap, the VelCrossover parameter does not appear on the Keymap-editor page. When the current keymap is dual-velocity, the value for the Crossover parameter will be one of the eight dynamic markings from ppp to fff. The K2600 translates each of your Note Ons into one of these dynamic values, using the settings for the VelTouch or VelocMap parameters. When this translated value exceeds the setting for the VelCrossover parameter, the K2600 plays the sample assigned to the upper velocity range.

When the currently selected keymap has three velocity ranges, the VelCrossover parameter becomes two parameters: LowCrossover and HiCrossover. The K2600 plays the sample assigned to one of these ranges depending on the translated value of each note's attack velocity.

When you want to use a keymap with more than three velocity levels, pick one with velocity crossover levels that you like, since you can't change the crossover velocity in keymaps with more than three levels.

## Building a Keymap

If you used the Keymap Editor to enter the SampleMode page, then just press **Exit** from the SampleMode page and you are ready to begin creating a keymap. If you entered the SampleMode page from Master mode, do the following. Start in Program mode, and select Program 199, the Default program. Press the **Edit** button, and you'll enter the Program Editor. Press the **KEYMAP** soft button, and the KEYMAP page will appear. The Keymap parameter will be automatically selected. Press **1, 6, 8, Enter** on the alphanumeric pad to assign the keymap **Silence**. This isn't absolutely necessary, but it makes it easier to recognize the key ranges that have samples assigned to them when you start assigning samples. You can actually choose any program you want to start with, but by choosing these, you are starting with a "blank slate."

With the Keymap parameter still selected, press the **Edit** button, and you'll enter the Keymap Editor. The Key Range parameter will be automatically selected, and you see its values: C 0 to G 10 (the entire MIDI keyboard range). The Sample parameter will have a value of **168 Silence C 4**.

Now you're ready to start assigning samples to key ranges within the keymap. We'll assume that you've loaded samples with roots at C 1, C 2, C 3, etc. and that you plan to assign a root to each octave. To begin, press the **Assign** soft button. The display will prompt you to select a sample. Use the Alpha Wheel to scroll to one of your samples, or type its ID on the alphanumeric pad and press **Enter**. When you've found the sample you want to use, press the **OK** soft button. The display will say "Strike low key..." Trigger A 0 (MIDI note number 21, the lowest A on a standard 88-note keyboard). The display will change to say "Strike High Key..." Now trigger F 1 (MIDI note number 29). The display will return to the Keymap-editor page. The Key Range parameter will show A 0-F 1, and the Sample parameter will show the sample you selected when you started the range assignment.

One more time... Press the **Assign** soft button. Select another sample root at the prompt, and press the **OK** soft button. Now trigger F# 1 for the Low Key prompt, and F 2 for the High Key

prompt. At this point you've defined two key ranges, the first from A 0 to F 1, and the second from F# 1 to F 2. You can repeat the process as many times as you want, creating a new key range each time.

Once you have your samples assigned, you may need to transpose them so that they play back at the correct pitch within the range you have chosen. To do this, highlight the Key Range parameter, scroll to the range you need, then highlight the Coarse Tune parameter. Adjust Coarse Tune to bring the sample to the proper pitch within that key range. Then scroll back up to the Key Range parameter, select the next range, and continue as needed.

Here's a fairly important point that may or may not affect your keymap construction. Suppose you want to build a keymap that uses the same sample in several adjacent key ranges, and you plan to add a bit of detuning to the samples in each range. You might think that you could build the keymap first, then go into the Sample Editor and tweak the samples when the keymap is finished. Yes, but...

Suppose you used the technique we described above to assign a vocal sample whose root was C 4 to a key range from A 3 to E 4. Then you assigned the same sample to a key range from F 4 to B 4. You might be surprised to find that when you finished the F 4–B 4 key range and the Keymap-editor page reappeared, the current key range would not be F 4 to B 4, but A 3 to B 4! This is because the K2600 automatically merges adjacent key ranges that are identical (this is done to save memory). Therefore, some parameter must be different in each adjacent key range you create if you want to build keymaps using the technique we just described. So if you want to use the same samples in adjacent key ranges with, for example, minor pitch or volume modification, you should make those changes to the current sample on the Keymap-editor page *before* assigning the next range.

## Using the Analog Inputs to Trigger Samples

The analog sampling inputs double as one- or two-channel trigger inputs. This allows audio signals from external sources (such as microphones and tape recorders) to trigger internal samples. The following steps explain how to use the trigger inputs.

1. Connect cables from the outputs of your external audio source to the analog sampling inputs on the K2600. For details on which cables to use with the analog sampling inputs, read *Setting Up For Sampling* on page 14-1.
2. Enter Program mode, and select the program containing the sample(s) you want to trigger.
3. Set the Song-mode Click parameters to match the program you chose:

Enter Song mode, select the MISC page, and set the Click Channel and Click Program parameters to the MIDI channel and program number of the program you just chose. Change the value of the Click Key parameter to define the key number played by the sequencer's metronome. This indirectly determines what samples are triggered by the inputs, since the inputs always trigger the samples that are assigned to the first two keys above the Click Key.

The default Click parameters are Click Channel **16**, Click Program **198 Click**, and Click Key **C4**. (Therefore the default keys triggered by the analog inputs are C# 4 and D 4.)

4. Enter Sample mode, set Src to **External**, and set Mode to **Trigger**.
5. Set the trigger threshold:



Highlight the **Thresh** parameter on the **SampleMode** page, and adjust the input sensitivity so that your input signal causes smooth triggering. Threshold values approaching **-90db** let you trigger the sample using a lower input level, while values approaching **-6db** require a greater input level.

Now analog input signals will trigger samples in the click program. The sampler's analog inputs are stereo, so you can trigger samples on two different keys. The left channel will trigger samples that are one key above the **Click Key**; the right channel will trigger samples that are two keys above the **Click Key**.

One great use for the trigger inputs is to replace the kick and snare from a multi-track recording with samples in your K2600. Assuming the kick and snare are recorded on separate tracks, and the **Click Key** is **C4**, the recorded drums will trigger samples assigned to **C#4** and **D4** in the click program.

Keep in mind that the **Click Key** number is separate from the trigger key numbers. This is because you may want to use the click program as the metronome sound, and trigger samples other than the **Click**. To do this, go to **Program** mode, select the click program, and press **Edit**. Then press the **KEYMAP** button and press **Edit** again. Now assign your samples to the two keys above the **Click Key** defined on the **Song-mode MISC** page. Before you leave the **Program Editor**, be sure the **KeyTrk** parameter on the **KEYMAP** page is set to **100ct/key**.

When you save the edited click program, remember to replace it in the same location. If you save to a new location, set the new program number in the **Click Program** parameter on the **Song-mode MISC** page.

## Live Mode

If you have the sampling option, you can use what we call **Live mode**. In **Live mode**, the K2600 takes any input signal and routes it through the **VAST DSP** algorithms and **KDFX**. You can connect any audio source—synths, mics, CD players, anything—to any of the K2600's sampling inputs, and treat that input as if it were a regular **VAST** program.

The easiest way to use **Live mode** is to use one of the factory programs (**740–749**). Some of the programs are optimized for certain applications (for example, guitar cabinet simulations), while others are meant to be used as templates.

You can't use **Live mode** and make samples at the same time, since both features use the same internal components.

### Creating a Live Mode Program

1. Press the **Sample** soft button to bring up the **SampleMode** page.
2. Set the **Src** parameter for the source you are using.

For example, if you have plugged a microphone into the K2600's **HiZ** sampling input, choose **Ext**. Be careful if you choose **Int**, since you can inadvertently create a feedback loop.

3. Set the **Mode** parameter to **LiveIn**.

Two samples will automatically be created: **197 Live Input L** at **C 4** and **198 Live Input R**, also at **C 4**. The soft buttons on this page are disabled when you set **Mode** to **LiveIn**.

4. Use one or both of the live-input keymaps (**197** and **198**) in an existing **LM** program, or in one you create.

For a stereo program, set Stereo to **On** on the KEYMAP page in the Program Editor.

5. Edit the program's parameters for the effect(s) you want to use.
6. Play C 4, then input the audio source that you want to run through Live mode.

Hint: Set VelTrk on the EditProg F4 AMP page to **0 dB**; otherwise the velocity with which you strike C 4 will affect your output. For alternative ways of triggering the sound (for example, with assignable controller buttons or pedals), edit the control setup.

You should now hear your VASTed audio source through the K2600's Mix outputs.

## Live Mode Programs

ID	Program Name
740	LM VirtualDesk 1
741	LM VirtualDesk 2
742	LM EQ Room Hall
743	LM TubeAmp_ Gtr
744	LM Synth Sliders
745	LM EQ Stlm Hall
746	LM ParaFlange
747	LM EQ Overload
748	LM Filters
749	LiveMode Default

Live mode also includes two Live mode keymaps at 197 and 198 (Left and Right respectively).

### Usage Notes

To use the programs, you must hold down a key (C 4, unless you're going for a special effect) for the inputs to run through VAST. An alternative way to trigger the sound is to edit the control setup found in the MIDI-mode TRANSMIT page. For example, on the SWITCH page in the Setup Editor, you could set the switch type (SwType) to note toggle (Note T), and set the destination (Dest) to C 4. This allows you to turn the program on and off via a button press, and keeps sound sustaining while the button is on. Rack-model owners can get the same results with any programmable switch controller. Keep in mind that if you change the Live mode program, you need to restrike a key (or button) for the signal to go through that program.

You can also edit the Live mode keymap to ignore release if you want to use the keyboard to activate Live mode.

You cannot sample and use Live mode together, the two functions use the same components.

## Some Ideas for Using Live Mode

If you've ever used an old-fashioned mono analog synthesizer with an audio input (anything from a Moog Rogue to an ARP 2500 or Serge Modular), you know how much fun it can be to pass a musical signal through the synth and modify it in real time with the filters, envelopes, modulators, etc. Live mode brings that concept to digital synthesis, and lets you use all of the power of the K2600 on any kind of input signal.

For starters, you can simply hook up a CD player to one of the K2600's sampling inputs, get a bunch of your favorite CDs, and start fooling around. (A turntable works well too.) Here are some ideas for going further:

### Pitch Changing

Unlike an analog synthesizer, the K2600 makes it possible to alter the pitch of the incoming signal in real time. But the K2600 is not a conventional pitch shifter, so if you are used to working with such a device you will have to alter your thinking a little.

For example, when you bend the pitch down from the unity pitch (C 4), using a VAST function, it slows the playback of the incoming signal, but it doesn't change the rate at which the signal is coming in—your CD is still spinning, and putting out a constant audio signal. So as you lower the pitch, the playback lags behind, and when you return the pitch to normal, the playback snaps back to the present—which means some of your audio literally disappears into the ether. If you bend the pitch down and hold it there for a while, eventually the buffer fills up and updates itself, and you will hear it snap forward in time, although the data playing will continue to be slowed down. Again, some of the audio disappears.

When you bend pitch upward, the K2600 plays buffered data from the input source, which enables the K2600 to “play ahead” of the input. You may hear some of the input data repeat when you release the pitch bend.

These details aside, all kinds of wonderful pitch effects are achievable. Here's an example.

1. Start with Program **749 LiveMode Default**.
2. Go to the PITCH page.
3. Assign LFO1 to **Src1**, with a depth of **-200ct**.
4. In order to keep the playback from constantly crossing above unity, set the Coarse parameter to **-25T**.

Or try these settings:

Src1     **MWheel**

Depth    **-1200c**

Src2     **LFO1** (On the LFO page, set LFO1's MnRate to **.50Hz**, MxRate to **20.00**, and RateCt to **Data**.)

DptCtl   **MWheel**

MinDpt   **0ct**

MaxDpt   **1200ct**

Sometimes the Live-mode audio will sound discontinuous as LFOs and the buffers get out of sync. You might be able to smooth out the rough spots by making another layer with no pitch alterations, and crossfading between the layers:

1. Duplicate the layer.
2. Clear all the settings on the PITCH page.
3. Go to the OUTPUT page and set Crossfade to **MWhl** on both layers.
4. On layer 1, set XFadeSense to **Rvrs**; on layer 2, set XFadeSense to **Norm**.

Now at the Mod Wheel extremes, you will hear only one layer or the other, while in the middle, you will hear a combination of the pitch-modulated signal and the unmodulated signal. By experimenting with FUNs, you can get more precise crossfades.

The program **744 LM Synth Sliders** includes this kind of crossfade, tied to the Pitch Wheel, to implement a 3-layer crossfade. Moving the Pitch Wheel up fades to a layer which is bending the pitch up. The surprise is that moving the Pitch Wheel down bends the pitch down, then up again, crossfading to a layer that is playing back in reverse! Yes, reverse playback works with Live mode: on the KEYMAP page, set PlayBackMode to **Rvrs**.

### Arpeggiator

You can also do controlled pitch shifting on incoming audio using the arpeggiator. By constantly sending new note starts, it is possible to bend the pitch without losing the tempo of the incoming signal.

It can work in both directions, although when you are shifting signals up in pitch, you're "borrowing" the audio from a few seconds previous.

1. Go to Setup mode and select **97 Control Setup**.
2. Press **Edit**, and on the CH/PRG page, set the program to **749 LiveMode Default**.
3. With the program highlighted, press **Edit** and go to the AMPENV page.
4. To make the crossfading less choppy, you want short attack and release segments: set Att1 to **0.06/100%** and Rel1 to **0.10/0%**.
5. Press **Exit** and save the program to some new ID.
6. Now go to the ARPEG page and set the Active parameter to **On**.
7. Set the Duration parameter to **99%**.
8. For this example, set Order to **Simultaneous** and Beats to **1/32**.
9. Tempo should already be **120**.

Now play C 4 and you'll hear the live signal at the correct pitch. Play G 3 and you will hear the signal pitched down a fourth. You can use the ribbon or similar controller to bend the pitch smoothly. Going above unity pitch will cause a jump back into the past.

Experiment with the Tempo, the Beats setting, the Duration value, and the AMPENV parameters to get useful variations on the program. Remember that because we set the Order to **Simultaneous**, you can play several notes at once. And finally, try setting Glissando to **On**.

### Sustained Notes and Loops

If the incoming signal is a single, sustained pitch, like a saxophone note, then you can consider the Live mode keymap to be playing a normal, looped sound. In this case, the fact that an upward bend jumps back a few seconds is no big deal because the sound hasn't changed much during that time.

With this technique, melodies or chords can be played based on a segment of a live performance. Keep in mind that, unless your incoming signal is a C, notes and chords played on the K2600 keyboard will be transposed relative to the incoming pitch. Also remember that a rhythm pitched an octave down will play at half the speed, while one pitched an octave up will play twice as fast. Fifths produce a 3-against-2 pattern. To keep some sort of relative sync with the

live signal, you may want to experiment with retriggering the notes, perhaps using the arpeggiator, at some appropriate tempo.

If the passage you want to play is long, and the input signal isn't so long—say, the sax player needs to take a breath—you may run into a problem as the K2600 tries to play the buffer where the audio was interrupted. If the input signal is mono, you might be able to overcome this by using a delay line to “hold” the signal. The delay line could be part of VAST, or it could be an external device, but either way its output is sent back to the K2600 through the unused Live mode input channel.

### Chord Progressions

Record a few bars of block chords—all notes under C 4—into the sequencer, using a simple quarter-note or half-note pattern. What sound you use doesn't matter. Now replace the program on the recorded track with the Live mode default program. Play back the sequence (you will probably want it to loop), and at the same time play single notes from an external instrument into the K2600, at the same rhythm as your recorded chords. If you change the notes on the instrument, the chords will transpose. If you play intervals or chords, you're on your own as to the consequences!

### Feedback

Live mode gives you the ability to feed back a live signal into itself, similar to pointing a microphone at the speaker it's sending audio to. Before you hook anything up, turn the volume down as low as you can.

Now go to the Sample page and set Source to **Internal**. Go to a multi-layer ROM program of your choice, and go to the Import page. Import Layer 1 from the Live-mode default program.

Play one note, then a few. As you play more notes, the noise will build up. You'll have a better time controlling the feedback loop if you have a healthy delay, with no dry path around it, in the loop. Perhaps add a little modulation of the loop to provide some pitch shifting, a big reverb, and a compressor to keep from blowing your ears out. Inject a little something from the synthesizer to get things started—and you are instantly transported to an alien dimension.

For more complexity, split the incoming signal and run it through multiple VAST layers in parallel—you can use up to 32, each one processing, panning, and routing the signal differently. You can crosslink the inputs and outputs (right into left, left into right) to create a double feedback loop for even more fun.

