

Chapter 15

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 K2vx. 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 K2vx Sampling Option.

Cables and Input Jacks

If you're going to be sampling from an analog source, connect a standard 1/4-inch mono or stereo cable (a typical guitar cable) from the output of your sample source to the 1/4-inch analog input jack on the K2vx. If you have a K2vxR or K2vxRS, use the 1/4-inch jacks if you're sending an unbalanced signal, or the XLR jacks on the front panel if you're sending a balanced signal. Although it is 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 will send the signal to the K2vx's left channel. If you use a mono cable, be sure to set the Mode parameter on the Sample mode 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 K2vx accepts a standard male XLR fitting. The K2vx'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 (or a cable and adaptor) 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.

If your digital sample source has an optical output, connect your cable to the optical input jack above the coaxial jack on the K2vx'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 - Two Different Ways

There are two different methods of entering the Sampling page. Which method you choose depends on the type of sampling you are doing - how many samples you are making and if you need custom keymaps.

The difference between the two methods primarily concerns the ease of accessing the keymap editor. Once you have made your samples, you will need to have them assigned to a keymap and have that keymap assigned to a layer in a program. Refer to the section entitled "Building a Keymap" on page 15-37, for a step by step explanation of how to create keymaps.

From Master Mode

The simplest way to enter the Sampling page is from Master Mode. Press the soft button labelled **Sample** on the Master page. 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 will allow you to quickly create a program and keymap, with that sample assigned across the

entire range of the keyboard. The program is a one layer program which uses the settings from the Default program 199.

From the Keymap Editor

This is a better method to use if you are 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 will jump you to the Sampling 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 which uses your new keymap, or you can return to the Sampling page for another round of sampling

Sampling Analog Signals

The K2vx'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 Sample mode 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. 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 K2vx 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 K2vx. The best results will be achieved by sampling from a digital source, using one of the K2vx's digital sample inputs.

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

Input

On the Sample mode 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."

```
SampleMode Samples:8192K Memory:752K
Sample:None Src:Int
Input :Analog Time :1s Mon:Off
Gain :0 dB
Rate :44.1KHz
Mode :Mono(L)
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 (Rack models only)

The K2vxR and K2vxRS have an additional parameter related to input: 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 K2vx's sampling inputs. Use a value of Int if you want to sample the K2vx's own output. (This can be done on the keyboard models of the K2vx by selecting Analog input and removing any cable connected to the analog sampling input). For sampling the K2vx's own output, the Monitor parameter must be Off to prevent signal attenuation.

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 K2vx'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 K2vx 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.

The meters are inoperative during actual sampling, so make a few tests of your levels before you begin to record.

Rate

After you've set your levels, you need to select the sample rate. You have four rates to choose from. The trade-offs that determine your best sampling rate are frequency response and storage requirements. Higher sample rates will capture more frequency content from your samples, but will 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 will 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 K2vx's transposition range during sample playback. The K2vx transposes samples by changing the sample playback rate; the higher the playback rate, the higher the pitch of the sample. The K2vx can achieve a maximum sample playback rate of 96 KHz. A sample made at 48 KHz can be transposed up only one octave, since the playback rate doubles for every octave of upward transposition. A sample made at 29.4 KHz can be transposed up approximately 21 semitones (an octave and a sixth). There is no limit on downward transposition, regardless of the sample rate.

Each portion of a sample (each individual sample element made by the K2vx 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 K2vx. 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 SIMMs for Sample RAM" in Chapter 8 of the *K2vx Reference Guide* before you go SIMM shopping, though.

At a sampling rate of 44.1 KHz, each megabyte you add increases your sample time by about 12 seconds. The chart below lists a few standard sample RAM configurations and their total sample time capacity (in seconds) at various sample rates.

		Sampling rate in KHz			
		29.4	32.0	44.1	48.0
2 x 1M	Mono	35	32	23	21
	Stereo	17	16	11	10
2 x 4M	Mono	140	128	92	84
	Stereo	70	64	46	42
2 x 16M	Mono	560	512	368	336
	Stereo	280	256	184	168

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(L) 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 Sample 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.

Threshold (Thresh)

The Thresh parameter controls when the K2vx 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 K2vx 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.

The K2vx has a pre-trigger feature—that is, the K2vx records the 3,000 samples immediately before the point at which the threshold is exceeded. This prevents early transients from being missed. This feature is automatic whenever the Thresh parameter is set to a non-zero value. You'll want to adjust this parameter if your samples are missing these transients; the lower the threshold, the more certain you'll be of capturing every element of the signal. In some cases, you may get better results if you set the Thresh parameter to a value of Off, then press the

Record soft button and wait briefly before beginning the sound to be sampled. This will guarantee that every element of the sample is captured. After you've recorded the sample, you can always adjust the starting point on the TRIM page, to remove any silence before the start of the actual sample.

Sampler recording can also be triggered via the keyboard. On the Sample page, set Thresh to "key", then press Auto. Striking a MIDI note event now will trigger the sampler, set the sample placement on the keyboard and relative tuning 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 K2vx calculates this automatically, and limits the maximum value of the Time parameter accordingly. At a value of zero for this parameter, the K2vx will not record.

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 manually create keymaps and programs for them. With a value of None for this parameter, the K2vx 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 K2vx automatically creates a temporary keymap and program, based on the settings for Default 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 Sample mode 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 below called "Recording Samples."

If you don't have enough free program RAM, you may be unable to audition samples, since the K2vx 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 K2vx'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 Sample mode 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.

For sampling the K2vx's own output, the Monitor parameter must be Off to prevent signal attenuation.

Recording Samples

Pressing the Record soft button will 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 K2vx 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.

You can abort the recording of a sample by pressing the -/+ , 0 and CLR buttons simultaneously.

When recording is complete, the K2vx 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 K2vx 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 K2vx 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 Sample mode page. If you press **Yes**, you'll see the normal Save dialog. When you've saved the sample, you'll return to the Sample mode 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. This will begin sampling immediately (or when the sample threshold is exceeded if you have the Thresh parameter set to a value other than Off), and will return you to the Sample Recording page when the sample is complete. The root key is automatically assigned at C 4, and the sample is automatically saved at the next available ID above 199.

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 will start immediately after the LEDs flash. If you have the Thresh parameter set to a value other than Off, sample recording will begin when the countdown has finished and the threshold level is exceeded.

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 Default program 199 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 K2vx will create a keymap and a program, using the lowest available ID numbers in that bank for both the keymap and the program. The display will tell you the ID number of the new program.

Sampling the K2vx's Output

You can sample the K2vx's own sounds when in Analog sampling mode. If there are no cables connected to the Analog sampling input, the K2vx will record its output automatically. To do so, set the Src parameter on the Sample mode page to a value of Int. Then, just press the **Record** soft button and start playing. The K2vx will sample the note(s) as played, without any controller information (just the raw sounds).

Sampling Digital Signals

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

You'll notice that the Sample mode 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.

```

SampleMode  Samples:8192K  Memory:118K
Sample:None
Input :Digital  Time  :1s
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 K2vx 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 K2vx's optical input; this plug must be removed before you can connect the optical cable to the K2vx. 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 K2vx 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 15-1.

Format

Use the Format parameter to tell the K2vx 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.

When you begin recording, a box appears in the display that indicates that the K2vx has locked on to the digital signal. You'll see the word "LOCK," and the rate at which the signal was sampled will be shown. The K2vx automatically sets itself to this rate. If you don't see this box, check to make sure that the signal is getting to the K2vx.

The K2vx may not display the rate when you're using a rate other than the frequently-used rates of 32, 44.1 or 48 KHz. This is normal, and although the rate may not be displayed, the K2vx will still lock on to it.

Using the Digital Outputs

The K2vx features digital audio outputs with XLR connectors. There are two points to keep in mind if you wish to use them. First, the Mode parameter on the Sample mode page must be set to a value of Analog to enable the digital outputs. Second, the analog input must be disconnected from any cables.

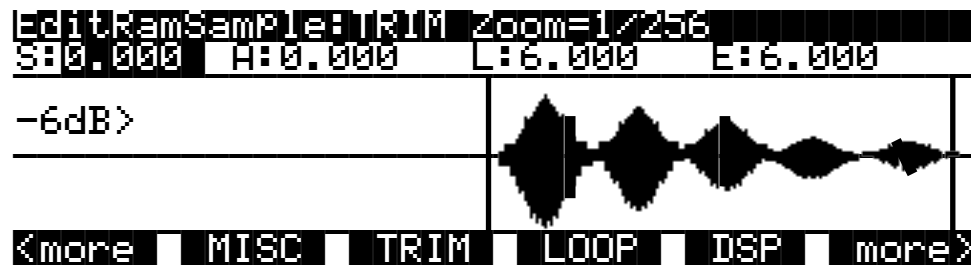
If you have a cable connected to the analog input, that input will be routed to the digital outputs. This effectively makes the K2vx into an analog-to-digital convertor, which can be very useful. The digital out will have the same signal that is sent to the MIX outs. So if you have all of your separate outs plugged in (and therefore have no signal coming out of the MIX outs), you will not have any signal coming out of the digital outs.

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, start by entering 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 K2vx 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. This button is available only when the currently selected sample is a stereo sample. When you press this button, the K2vx 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. 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.

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.

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 K2vx will ask you if you want to copy the sample data. If you answer Yes, the K2vx will make a separate copy of the sample. If you answer No, the K2vx 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 K2vx 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 K2vx, 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.

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).

Miscellaneous (MISC)

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   :G#1   LoopSwitch:On
PitchAdjust  :-12ct Playback   :Normal
VolumeAdjust :0.0dB  AltSense   :Norm
AltVolAdjust :-0.5dB IgnRelease:Off
DecayRate    :36B/s  SampleSize=101Kb
ReleaseRate  :106dB/ SampleRate=19200Hz
<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 (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 100 cts, 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 Chapter 6 for a discussion of AltSwitch.

Decay Rate

Use this parameter to specify how long the sample will take to decay (fade) to zero amplitude when a note is sustained, either by holding the key (or other note trigger), or with the sustain pedal. The higher the value, the faster the sample will decay. The sample starts decaying as soon as the note is triggered, and decays with a linear curve. This decay affects each sample individually, and begins as soon as the note starts. It operates independently of the decay in the AMPENV, which affects every note in the layer.

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 begins as soon as the note is released. It operates independently of the release in the AMPENV, which affects every note in the layer.

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 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; non-decaying 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.

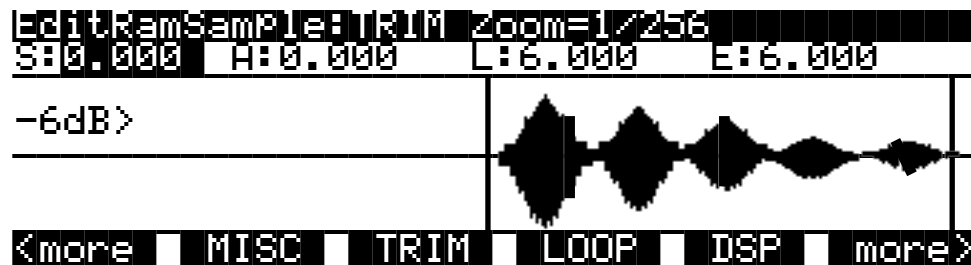
Sample Size and Sample Rate

These are not editing parameters. They're put 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.

TRIM

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 will overlap. Selecting one of the parameters will move the line corresponding to that parameter to the center of the display, where it will flash to indicate the parameter's position. The line corresponding to the currently selected parameter will flash. 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 be moved with it.

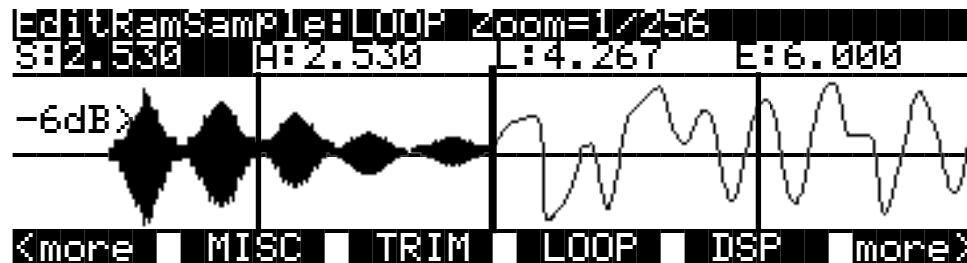
The End (E) parameter sets the point at which the current sample will stop playback. Typically you'll use this parameter to trim unwanted silence off the end of a sample, although you can use it to shorten a sample as much as you want.

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 save 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.

LOOP

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.

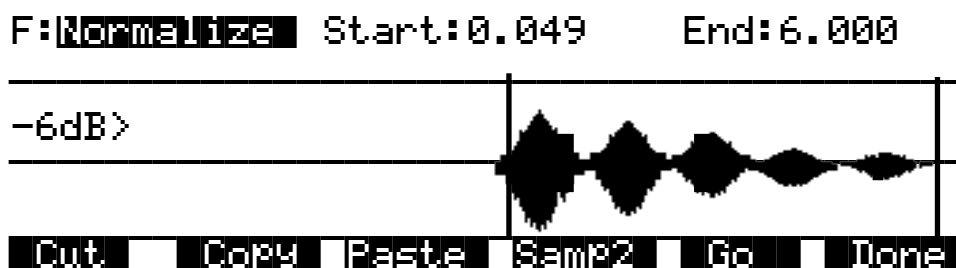
You'll also want to try to set the loop point at a zero-crossing—a point where the sample waveform is neither positive or negative. Pressing the Plus/Minus buttons together will search (from left to right) for the sample's next zero-crossing. You can usually press these buttons several times for any 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 K2vx will search for the next zero-crossing to the left. Press the Plus button again to search toward the right.

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.

DSP (RAM samples only)

Select the DSP page with the DSP soft button. This gives you access to a long list of non-real 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 K2vx 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 K2vx 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 K2vx 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 K2vx is shut off or reset (soft or hard). The buffer is also cleared if you execute a master delete, and select "Delete everything."

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 K2vx will prompt you to keep the changes. You can audition your changes and decide whether to keep them, then press Yes or No.

Done

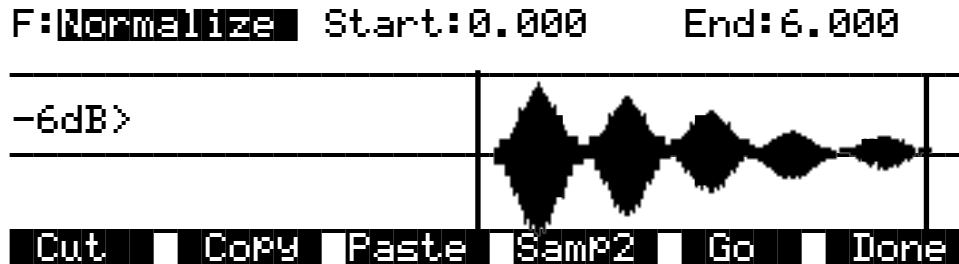
Press the Done soft button to return to the previously selected Sample Editor page when you're finished with the DSP functions.

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.

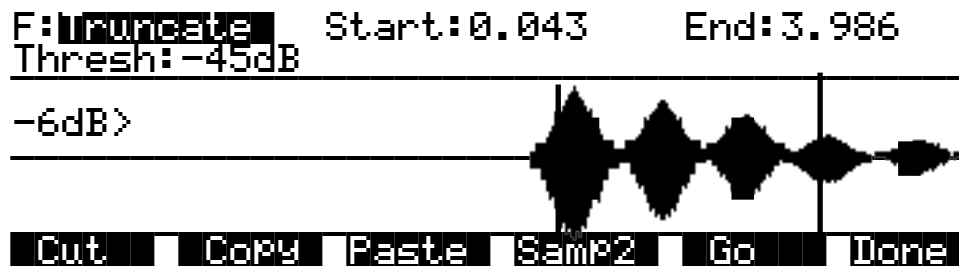
Remember that the DSP functions operate only on RAM samples. You'll notice that if you're editing a ROM sample, there is no DSP soft button; instead, there's an extra Link soft button for your convenience.

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 K2vx 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 Sample mode page, then press either the Save soft button, or EXIT.

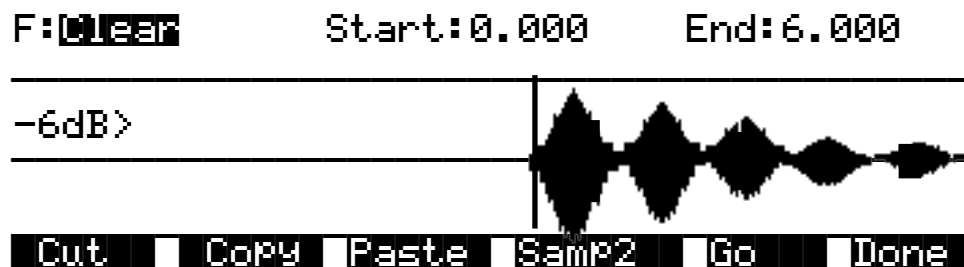
3 Volume Adjust



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 K2vx, 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.

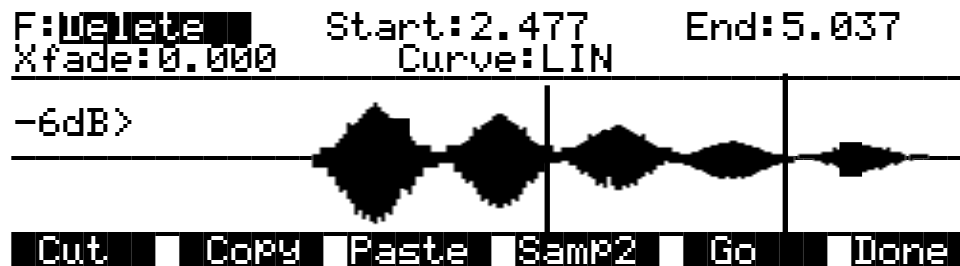
When you've selected the range to be adjusted, select the VolAdjust parameter with the cursor buttons, and use the Alpha Wheel to adjust the volume of the selected range. You can cut/boost the volume from -96 to 96 dB.

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



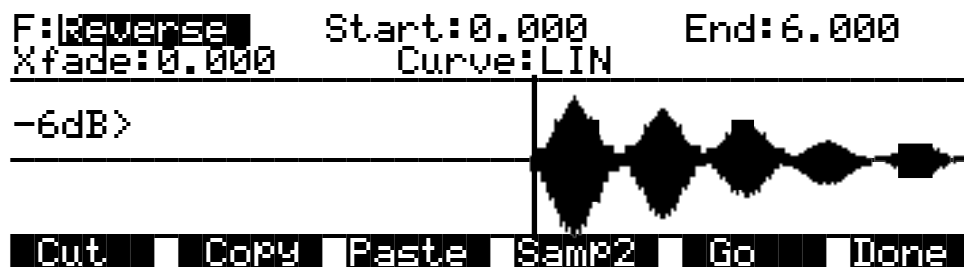
Unlike the Clear function, this will erase the samples within the selected range and shorten the sample, like cutting a section out of a tape and splicing the ends. If you want to silence a segment of the sample without shortening it, use the Clear function.

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.

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

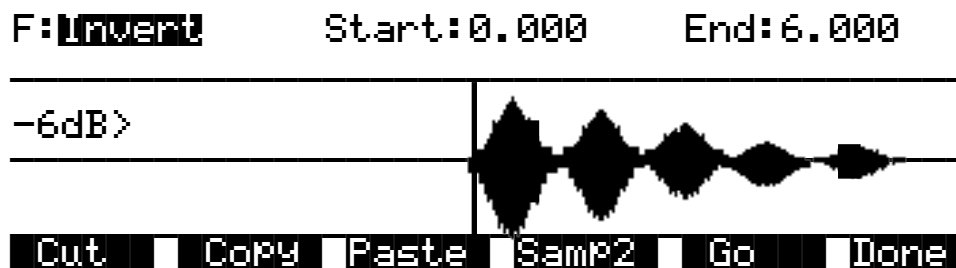


With this function you can reverse the order of the individual samples between the start and end positions you set. The Xfade parameter lets you apply a crossfade to the start and end of the reversed segment. The Curve parameter lets you select a crossfade curve. The available

values are LIN, EXP, COS, EQL, and MIX. These curves are described at the end of the section on DSP functions.

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 at the end of the section on DSP functions.

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 K2vx 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 keys 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 at the end of the section on DSP functions.

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 K2vx 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 at the end of the section on DSP functions.

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 at the end of the section on DSP functions.

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 K2vx 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 K2vx 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 an 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 ± 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. The most natural application of this function is to add percussion samples, but it can be used to mix any sample at regular programmable intervals (beats).

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. You can use the Incr parameter to shift the starting position of Sample 2. If you set a positive number, the mix range of Sample 2 will start later each time it's mixed. A negative value will mix sample ranges that start progressively earlier.

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.



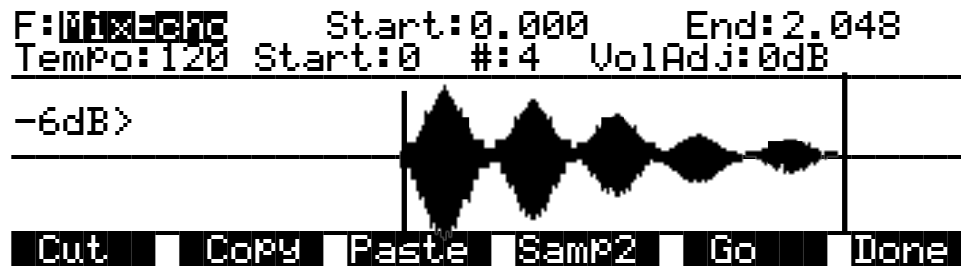
If you change Beat to 3, Sample 2 will play on beats 3, 7, and 11. If you set Beat to 1 and Of to 6, Sample 2 will play on beats 1 and 7. If you set Beat to -1, Sample 2 will still play on beats 1 and 7, but the sound of the sample will change each time, as the start point of the Sample 2 segment moves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

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.

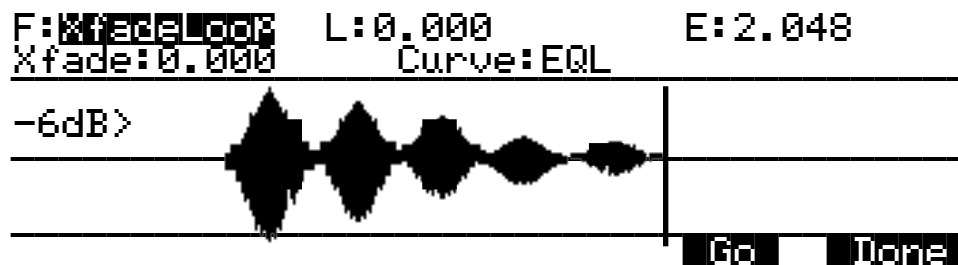
19 Beat Volume Adjust



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 lasting a half second, and bringing 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 below.

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 flat at both ends and steeper 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.



Reading Samples

The K2vx will load numerous samples from Akai, Roland, and Ensoniq EPS and EPS-16 Plus SCSI drives and floppies (including ASR-10 "Ensoniq" format), using Version 2's enhanced Disk mode operations. The displays you see will vary depending on the samples you're loading, but several features are the same. We'll describe the similarities first, then elaborate on the differences.

First, enter Disk mode by pressing the Disk mode button. Select the disk to be loaded from, as described in Chapter 13. The K2vx automatically recognizes the type of disk when you select it.

Press the Load soft button, and you'll see a page prompting you to select something to load (we'll call them objects, since different manufacturers give them different names). The top line of the display will tell you the number of objects available of the currently selected type, as well as the index number of the currently selected object. You can select any object in the list by typing its index number on the alphanumeric buttonpad and pressing Enter. The next step is to use the soft buttons to select the type of object to be loaded.

Once you've selected the type of object to load, press the OK soft button, and the bank dialog will appear, enabling you to select the bank into which the object(s) will be loaded. When you've selected a bank, press the OK soft button, and the loading process will begin. At the center of the display you'll see the object currently being loaded. The top line of the display will fill with asterisks to indicate the status of the current object. The bottom line will tell you the total number of kilobytes to be loaded.

The K2vx will create layers as necessary when you load objects. These layers have the same settings as Layer 1 of Program 199.

When the load is complete, the Disk mode page will reappear. You can now proceed with another load, or go to any other mode. If you exit Disk mode, the K2vx will remember the file that you selected most recently. When you return to Disk mode, this file will be selected.

Once you've loaded a sample or program file, you can save it as a Kurzweil object. You'll find it it can be loaded and backed up much faster as a Kurzweil object than in its original format.

Akai

The first page to appear is the page for loading files. The soft buttons name the operations: HDrive, Volume, and File on the left, and OK and Cancel on the right.

The hierarchy of objects is shown by the three soft buttons on the left. The display prompts you: "File to load:" The HDrive button selects the partition on the currently selected disk. The Volume button selects volumes within the currently selected partition. The File button selects an individual sample file from within a volume. The OK button, toward the right, executes the displayed function: partition selection, or loading a volume or file. The Cancel button returns you to the Disk mode page.

When you press the HDrive button, the center of the display's top line shows the currently selected volume in the currently selected partition. The prompt at the center of the display will read: "HD Partition." The list of available partitions will appear following the prompt. They're usually named A through F. Use the cursor buttons or numeric entry to highlight a different partition. Pressing the OK soft button will select the highlighted partition.

Pressing the Volume button will change the prompt to "Volume to load:" The list of available volumes in the current partition will appear. The center of the top line will show the current partition. The Layer buttons will scroll through the list of available partitions. Use the cursor buttons or numeric entry to select a different volume. Pressing the OK button will load the entire highlighted volume, unless the volume is larger than your available sample RAM, in which case, the K2vx will load as many files as will fit.

The Bank dialog will appear, enabling you to select the bank that will receive the volume. Press OK again, and you'll be prompted to press either the Progs soft button, which will load program information in addition to the samples, or the Samps soft button, which will load only the sample information. Programs are identified by the suffix ".p," and are stored in program

RAM. Samples have the suffix “.s,” and are stored in sample RAM. You can press Cancel to return to the Disk mode page without loading the volume.

If you load sample objects, you’ll see the following prompt: “Create preview program/keymap?” If you answer Yes, the K2vx will load the samples into a program that it creates based on Layer 1 of Program 199.

Loading program objects will load multi-layer samples and keymaps, and sometimes velocity switches. The K2vx will create layers as necessary when you load program objects. These layers have the same settings as Layer 1 of Program 199. In some cases, the K2vx will also create stereo keymaps to preserve the separation of stereo samples.

If you press the File button, the prompt will change to “File to load:” You can view the list of files with the cursor buttons, or use numeric entry. The top line of the display will show the currently selected volume. Select different volumes with the Layer buttons. The size of the currently selected file, in Kilobytes, is shown just above the soft buttons on the left. Press the OK button to load the highlighted file.

Press OK, and the Bank dialog will appear. Press OK again, and the file will be loaded into the highlighted bank.

Sometimes you may find an Akai floppy disk that your K2vx can’t read. In this case, check the format of the disk. Akai allows you to format a double sided-double density (DD) disk for high density (HD). This is not standard in the computer industry, since DD disks are not verified for use as HD disks. If this is the case, you will need to load the disk into an Akai, then save it out to a properly formatted disk.

Roland

For Roland disks, the hierarchy is a bit different; the objects that can be loaded are called Volumes, Performances, Patches, and Samples. The page that was selected last time a SCSI load was executed will appear when you initiate the load operation. Following the prompt is the list of available objects, with the size of the object in Kilobytes displayed as well. The top line of the display will show the number of available objects of the selected type.

Use the soft buttons to highlight the object to be loaded. The layer buttons will take you through the current object list in increments of 100. Press OK to execute the load. The Bank dialog will appear. Press OK again, and the object will be loaded. The display will update you on the progress of the load.

EPS

For EPS disks, the hierarchy consists of files and directories. Directories can be nested several layers deep. When you press the Load soft button, you’ll be prompted to select a file or directory to load from the list of available files and directories. The currently highlighted object will be either a file or a directory. If it’s a file, its name and size will be shown following the prompt. If it’s a directory, its name appears, followed by “(dir)” to indicate its type. The Layer buttons will take you to the first and last files of the currently selected directory. Note that if you’re loading from floppy disks, you won’t see any of the displays or soft buttons relating to directories, since the directory architecture is not supported for floppies.

When a file is highlighted and you press OK, the Bank dialog appears; press OK again to load the file. When a directory is highlighted and you press OK, you enter that directory, and the list of files and subdirectories in that directory appears, each file followed by its size, and each subdirectory, if any, followed by “(dir).” The top item in every list you select is always the parent directory of the files below it. Select the top item in a list to go up one directory level.

Pressing the Exit soft button will take you one level back up the hierarchy. Pressing it repeatedly will take you to the root directory—the directory at the top of the hierarchy. The quickest way to the root directory is to press the Root soft button. The top line of the display shows you the name of the currently selected directory (or subdirectory).

Pressing the All soft button will load all files in the current directory (but not any subdirectories). The Bank dialog will appear, and when you press OK, you’ll be prompted to

press the Progs button to load program information in addition to the samples, or the Samps button to load only the samples.

The Keymap Editor

The Keymap Editor lets you customize the K2vx'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 K2vx 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 K2vx 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 K2vx 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. 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 can not 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 K2vx, 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. The program Dual Electric Piano, for example, uses a keymap with two velocity ranges. Each key range in a multi-velocity keymap contains two or three distinct sample roots that the K2vx 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, start in Program mode by selecting a program that contains a multi-velocity keymap. Most of these programs are identified by the word “Dual” in their names. Then select the KEYMAP page, and press EDIT to enter the Keymap Editor. From there you can change the sample root assigned to each velocity level in each key range. The description of the Velocity Crossover parameter on page 15-36 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 15-37). 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-A 1  Lo:C 0  Hi:A 1
Sample         :1 Grand Piano-G#1
Coarse Tune    :0ST
Fine Tune      :0ct
VolumeAdjust   :0dB  VelCrossover:None
Name  Save  Delete  Dump  NewRng  Assign

```

PARAMETER	RANGE OF VALUES
MASTER TRANSPOSE	-128 ST -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 15-36.

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 K2vx 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.

The 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 Transmit page. If you change the value here, the same value will be reflected on the MIDI 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 which has fewer than 88 notes.

As with the Transpose parameter on the MIDI Transmit page, if you have the rack, this parameter will not transpose without using the Local Keyboard Channel parameter (refer to page 10-6 for more on the Local Keyboard Channel).

The transposition amounts from this page and the Master page are cumulative. For example, if you set this parameter to a value of +1ST and then set the Transpose parameter on the Master page to -1ST, the two will cancel each other out, and the K2vx will actually be back at 0ST.

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 K2vx (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. For RAM samples, each sample will have its own ID number. For ROM samples, it's a little different. Because of the large number of ROM samples, we did not want to use up too many of the available IDs. So we have organized them as follows. Each sample's name consists of three parts—for example, "1 Grand Piano G#1." The first numeral indicates what we call the sample block, which represents groupings of similar sample roots in memory. Next comes the name of the sample, which 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 key, then play a note on your controller 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.

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 (*Dual* Electric Piano, for example). Multi-velocity keymaps have a predetermined number of velocity levels, each of which can be assigned a different sample.

The K2vx supports keymaps with one, two, or three 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. Keymaps 2, 10, and 53 are dual-velocity keymaps. Keymaps 164–166 are triple-velocity keymaps.

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 K2vx 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 K2vx 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 K2vx plays the sample assigned to one of these ranges depending on the translated value of each note's attack velocity.

Building a Keymap

If you used the Keymap Editor to enter the sampling page, then just press Exit from the Sampling page and you are ready to begin creating a keymap. If you entered the Sampling 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) from your MIDI controller. The display will change to say "Strike High Key..." Now trigger F 1 (MIDI note number 29) from your MIDI controller. 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 Keyrange 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 keyrange. Then scroll back up to the Keyrange 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 K2vx 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.