EOS 4.01 Operation Manual Addendum

E-MU PN: FI10199 Rev. A

Introduction

This is the Operation Manual Addendum for EMU Operating System version 4.01.

EMU Systems' policy of continuous product support has led to the development of the EOS 4.01 Classic update. This update allows the E-IV, e-64, E4K, E-6400, E4X, and E-Synth to share the same software features created for our latest Ultra series samplers. EOS 4.01 Classic software has all the features of 4.0 Ultra software with the exception of the Beat Munger. *(Beat Munger requires the extra computing power of the Ultra series.)*
Nevertheless, EOS 4.0 Classic contains a host of great new features that you're going to love.

Important Upgrade Information

The features and functions of EOS 4.01 Classic require the expansion of the Flash Memory SIMM where EOS resides in your Emulator. Different Emulator models require different upgrades.

- **EIV, e-64, E-6400, E4X, E-Synth rack -** User installable - Use Kit 6853
- **E4K, E-Synth Keyboard -** Service Center Upgrade - Call for information

Flash Memory Notes

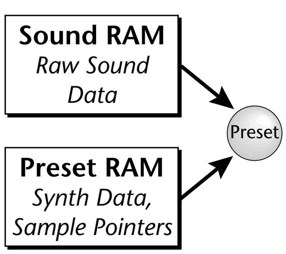
EOS-based Emulators have always contained Flash memory that can be used to store presets and sequences. However, this feature was not fully explained until the advent of the E-Synth, which utilized ROM-based samples and presets stored in Flash memory. A little known fact is that Flash memory can be used to store presets and sequences even if your Emulator does not contain ROM sounds. Since the 4.01 Classic upgrade has vastly increased the size of your Flash memory, you can now freely utilize this EOS "Power Feature". First though, a little background…

Five Types of Memory

The Emulator uses up to five different memory types. Yes, we know this *CAN* be confusing. Hopefully this will clear things up a bit.

- **Sample RAM -** 4-128 MB. Contains samples loaded from the hard disk.
- **Sample ROM -** Permanent ROM samples. Optional on all but E-Synth.
- **Preset RAM -** 1-8 MB of dynamic RAM. Contains the preset data and sequences which have been loaded from the hard disk.
- **CPU Flash -** Normally used to store presets for ROM or Sound Flash banks, but can also be used to store presets for HD based sound banks. However, this can be confusing since the proper samples may not be loaded into Sample RAM when the preset is selected. Sequences can also be stored in CPU Flash.
- **Sound Flash -** Optional flash SOUND memory to store your own banks.

Sample RAM & Preset RAM



EOS presets use two separate memory locations — Preset RAM and Sample RAM. **Sample RAM** contains the raw sound data. **Preset RAM** (also called CPU memory) is used to hold both preset parameter information and sequences. (The ratio between presets & sequences is user adjustable—*Master, Setup, Memory*). The amount of Preset Memory determines the number of presets and sequences you can have in your Emulator.

Preset RAM on the e-64, E-IV, E4K, and E-6400 is about half a megabyte. The E4X, and E-Synth come standard with 2.6 MB of usable preset RAM. Ultra models have about 4 MB of preset RAM. Sample RAM contains the actual sample data and can vary from 4 MB to 128 MB. Because presets use far less memory than do samples, less Preset RAM is needed.

Sound ROM & Preset Flash

Sound ROM is a standard feature on E-Synth and one of the available hardware upgrades for other EOS instruments. Although the samples themselves are in ROM and cannot be changed or modified, the Presets with all their keyboard mapping and synthesizer parameters *CAN* be changed because they are stored in non-volatile **Preset Flash Memory**. Flash memory contain 256 to 1000 presets depending on the amount of Flash memory you have installed. Flash presets are located immediately after RAM preset 999 (I000 - IXXX) in the preset numbering scheme.

Normally, preset data stored in Flash memory uses the samples stored in ROM. However, you can also create RAM presets (saved and loaded from disk) which use the permanent ROM samples.

Presets stored in Flash memory can also point to RAM based samples. **This mode of operation can be CONFUSING** however, because there is no guarantee that the correct samples will be in Sound RAM when the flash preset is selected. *(The Presets will play the wrong Samples.)*

- Banks of Presets are copied to and from a floppy or hard disk using the **Flash Memory Utilitie**s which are located under *Master, Bank, Flash*.
- Individual Presets are copied to and from Preset Flash using the **Preset Manage**, **Copy** utility. Select preset numbers (I000 - IXXX) for Flash.
- Sequences are copied to and from a floppy or hard disk using the **Sequence Copy Utility** which is located under *Master, Sequence, Manage, Utils*. Sequences stored in Flash memory have the prefix "z" and are located in locations z50-z99.

Tips for Using Flash Memory

Tip #1

Storing RAM based presets in flash can be useful to store preset "templates" that do not have samples associated with them. Bear in mind that unless you have Sound ROM or Sound Flash installed, the presets may NOT point to the correct samples.

Tip #2:

Flash presets can also be used when you're working on a RAM based preset which uses a lot of sample memory. Normally you would Save the bank, but this can take quite some time with a really large bank.

If you're only modifying the preset parameters and not the samples, you can Copy the preset to Flash and save time. Later you can Copy the preset back to RAM, then Save the entire bank normally.

Tip #3

Flash memory is also handy to store your favorite sequences. Copy the sequence to a Flash sequence location (designated by a (z) prefix).

New Features

Master Menu

Support for ADAT Digital Optical I/O

EOS 4.01 Classic software provides support for the 6861 ADAT Digital Optical interface card. The ADAT I/O card adds 16 digital output channels and 8 digital input channels. Since Emulators can only sample in stereo, only two of the 8 input channels can be used at once. The interface adds 8 digital output channels (sub 4-7) in addition to 8 digital channels which duplicate the 8 analog outputs (main, sub 1-3).

- Only Ultra models have the ability to be a master clock slave when using digital I/O. EIV, e-64, E-6400, E4X, E-Synth rack, E4K, and E-Synth Keyboard MUST provide the master clock to other digital devices.

ADAT Output Dither

This feature only appears when you have an ADAT I/O card installed.

Dither is a technique used in digital systems to improve audio performance by adding noise to the least significant data bits. In general, dither is used whenever a digital number is converted to a smaller number (for instance when converting 18 bits to 16 bits).

As an example, suppose you were transferring ADAT optical data from the Emulator to an older ADAT recorder. The Emulator ADAT output is 18 bits wide while the older ADAT only receives data 16 bits wide. In this case you would turn **Dither On** because the receiving device has fewer bits. If you were sending data to a new ADAT which receives 20 bit data, you would turn **Dither Off**.

When to Dither?
Add Dither *- When sending ADAT data to a 16 bit device which doesn't dither at the input.*

Don't Add Dither *- When sending ADAT data to a 20-24 bit device.*

☞ To Turn ADAT Output Dither On or Off:

1. Press the **Master** button. The LED illuminates and the Memory Statistics screen appears.
2. Press the **Setup** function key (F3). Another row of function keys appears.
3. Press the **Output** function key (F2). The Output screen appears.
4. Move the cursor to the **Output Dither** field using the cursor buttons or by pressing the F4 key.
5. **Turn Dither on or Off** using the Data Entry Control, or INC/DEC buttons.
6. Press the **Exit** button to return to the Memory Statistics screen.

Avoid Host on ID

This function tells EOS to avoid any host on the selected SCSI ID number. All Macs use ID 7. If you're a PC user, make sure this number matches your SCSI card.

This feature was formerly called, “Mac on SCSI Bus”. The Emulator can be connected via SCSI to the Macintosh, PC or SGI computers.

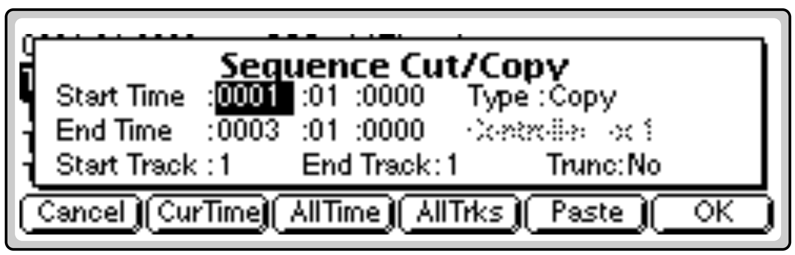
Improved Akai Conversion and Support for Akai S3000

The Akai sample and preset conversion routines have been substantially improved. They are now more accurate and convert more parameters. EOS now converts S3000 samples as well.

Sequencer Menu

Note-Off Truncate Feature in Cut/Copy

This feature allows you to selectively truncate Note-Offs which occur after the selected End Time. With Note-Off Truncation selected (Yes), all notes will be turned Off at the selected End TIme.



Yes

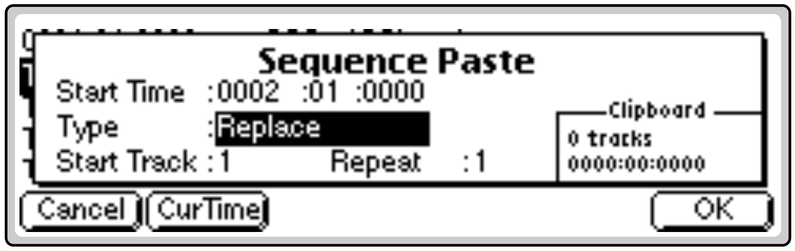
Any Note-Offs occurring after the selected End Time will be truncated to the selected End Time. The note length may not be preserved.

No

The Note-Off will occur as it was recorded, even if this time falls outside the selected End Time. The note length will be preserved.

Paste Repeat

You can now Paste multiple times, which is very handy for extending the length of a sequence.



Using the Repeat feature in Merge mode may clog your sequence.

Designed to be used in conjunction with **Insert** mode, Repeat Paste will Paste the clipboard contents from 1 to 127 times, lengthening the sequence.

☞ To Paste a Section:

1. From the main Sequencer screen, press the **Edit** function key (F5).
2. Press the **Utils** function key (F1). A second row of function keys appears.
3. Press the **Paste** function key (F2). The Paste menu appears.
4. Select the **Start Time** in bars/beats/ticks.

- Press the **CurTime** function key to select the current location as the Start Time.

5. Select the **Type of Paste**. Your choices are: Insert, Replace, or Merge.
6. Select the **Start Track** and **End Track** number.
7. Set the number of **Repeats** *(should be set to 1 except in Insert mode)*.
8. Press **OK** to paste the section or **Cancel** to cancel the operation.

Quantize Duration

Quantize changes the timing of the notes in a sequence so that they fall exactly on the beat. This is useful when your playing was sloppy or to create drum tracks. Quantization can be applied in note values ranging from 1/4 notes to 1/64th note triplets.

Duration (when checked) will also quantize the note-off to the nearest note value. This changes the note duration which would make a quantized quarter note exactly one quarter note long.

Note Erase

EOS contains a quick and handy way to erase notes from a recorded track. Only one note can be erased at a time. If more than one note is held, the last note pressed will be used for the erase.

Conditions: *For Note Erase to work…*

- You must be in **Record-Overdub** mode.
- **Loop Region** must be **On**.
- The sequence must be on the **second** or **subsequent loop**.

The **E** will appear in the display ONLY if the above conditions are met.

☞ To Use Note Erase Mode:

1. While playing back the sequence in **Record-Overdub** and **Loop** mode, **press** and **hold** the **Set/Shift key.** An "E" appears in the Edit screen.
2. **Play the note** you wish to erase on the keyboard.
3. **Release the note** to stop erasing.
4. **Release the Set key** to go out of Erase mode.

Sequencer Tips

Tip #1

If the metronome preset has effects on it (and you don't want effects):

1. Copy the preset.
2. Assign it to a Submix out with 0% effects.
3. Save the Bank.
4. Assign the Metronome to the new Preset.

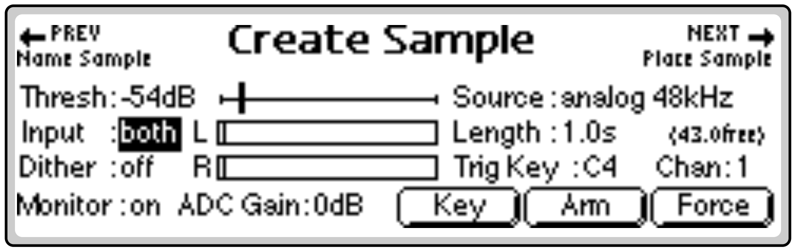
Tip #2

The Arpeggiator and Sequencer use the SAME CLOCK.!
Change the Arpeggiator clock to adjust the tempo of Flash sequences …
… or Copy the Flash sequence to RAM, then adjust the tempo.

Sample Manage Menu

New Input Channel Select

The Input Channel selector is now a single field which allows you to select Left, Right or Both channels. Formerly, each channel had its own On/Off control.



Select Left, Right or Both: The Input field determines which channels are On or Off and controls both the analog and digital inputs (AES and ADAT). The input levels are displayed in the VU Meter (the VU meter is used to help set the ADC Gain). Use the INC/DEC keys or Data Entry Control to select Left, Right or Both channels. If sampling in mono, use the LEFT channel ONLY since the Threshold trigger only monitors the left channel.

Sampling Source & Rate

The Source field defines the sampling source and the rate at which the information is sampled. It is important to use the lowest rate possible (to conserve memory) that is at least twice as high as the highest frequency component in the sound being sampled. If the sampling rate is too low, you will lose some of the high frequency information.

Analog Sampling: Choose between 22.05 kHz, 24 kHz, 44.1 kHz or 48 kHz when using the analog inputs. The sample rate should be at least twice as high as the highest frequency component in the sound being sampled.

If an ADAT Optical card is installed, you will have four additional digital sampling source options:

- *ADAT in 1 and 2*
- *ADAT in 3 and 4*
- *ADAT in 5 and 6*
- *ADAT in 7 and 8*

Digital Sampling. Choose between 32 kHz, 44.1 kHz or 48 kHz when using the digital input. The display setting should match the sample rate of the incoming digital data, otherwise the Arm and Force Sample function keys cannot be selected. If a digital source is not connected, the AES clock indicator displays “no AES” and sampling is disabled.

A sample overload message is displayed if digital audio containing “Full Code” data (all 1's) is detected. This indicates that the input data has been clipped at some point. Full code data may distort in the Emulator because changing the pitch can cause a slight gain increase. If possible, reduce the level of the incoming digital data if you see an overload message.

Resampling: This function allows you to resample the main outputs as you are playing. Resampling lets you build up extremely dense sounds or record a sequence of notes to be played back with a single key. There are two resampling modes, 16 bit and 18 bit. Use 16 bit mode for monophonic or duophonic passages. The 18 bit mode gives you more headroom when playing many notes at once. If clipping occurs, use a higher bit mode.

☞ To Set the Sample Source:

1. Press the **Sample Manage** key. The LED illuminates and the main screen appears.
2. Press the **New** function key (F3). The "Create Sample" screen appears.
3. Move the cursor to "**Source**" using the cursor keys.
4. **Select the desired sampling source** using the Data Entry Control or INC/DEC keys.
5. Press the **Exit** key to return to the Sample Manage screen.

Dither

Dither is a technique used in digital systems to improve audio performance by adding noise to the least significant data bits. Dither should be used whenever a digital number is converted to a smaller number (for instance when converting 20 bits to 16 bits).

EOS samples at 16 bit linear, so dither should be turned On when digitally sampling from a source that has more than 16 bits, for example when sampling a 24 bit AES signal or when sampling from a 20 bit ADAT machine. Dither can be either On or Off when analog sampling (the audio performance is the same).

Export Sample

This function allows you to export a single sample as a self-contained bank in any of four different sample formats. The formats are:

- **EIIIx** Compatible with EIIIx samplers
- **Emulator IV** Native format of all EOS based Emulators
- **WAVE** Microsoft Windows .WAV file format *(floppy export only)*
- **AIFF** Apple sound file format *(floppy export only)*

Suppose you were experimenting and modified a sample using the sample edit module. The sample data is permanently changed. You want to keep the modified sample, but if you save the bank, the original sample will be lost. Using this function, you can export the modified sample to a new bank and have both!

Hot Tip! *To save all the samples in a bank in EIIIx format, press and hold the decimal point key while you press the Save key. The display will offer you the option to save in these alternate formats. Select the desired format using the function keys.*
Note that only samples are transferred, not the presets.

☞ **To Export a Sample:**

1. Press the **Sample Manage** key. The LED illuminates and the main screen appears.
2. **Select the sample** to be exported using the Data Entry Control, INC/ DEC keys, or the numeric keypad.
3. Press the **Export** function key (F5). The screen shown above appears.
4. **Select the destination** drive, folder and bank using the cursor keys and Data Entry Control. **WAVE and AIFF formats can ONLY be exported to floppy disk.** You cannot write into an existing bank without erasing it. Usually you will choose an empty bank.
5. Select the format for the samples. Press **F2** for **EIIIx**, **F3** for **EIV** (EOS), **F4** for **WAVE**, or **F5** for **AIFF format**.
6. Press **OK** to export the sample or **Cancel** to cancel the operation.

Another New Feature!

Copy Sample now allows you to Copy from the Sample Clipboard!!!!

Sample Edit Menu

New Menu Ordering

The Sample Edit module has been reorganized into six main submenus.

Utilities. Cut, Copy, Paste, Truncation and Taper

Tools 1. Looping Controls, Loop Type, Digital Tuning, Sample Rate Conversion, and Sample Calculator

Tools 2. DC Filter, Swap Left and Right channels, Stereo<->Mono conversion, Reverse Sample, and Sample Integrity

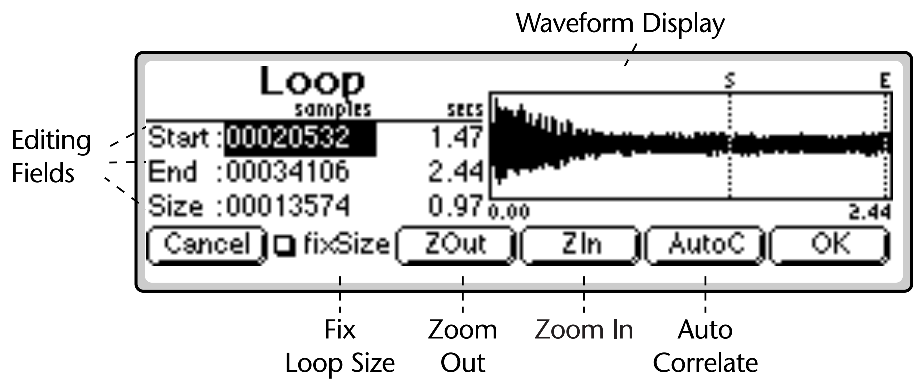
Tools 3. Gain Change, Compressor, Parametric EQ, FIR (Phase Linear) filter, and Aphex Aural Exciter™

Tools 4. Transform Multiplication, Doppler/Pan, Time Compression/ Expansion, Pitch Change, and Bit Converter.

UNDO! Allows you to undo any sample editing process.

Fix Size Looping Parameter

This feature was implemented by popular request. It gives you the option to either lock or unlock the loop size. You can still loop the way you always have by turning Fix Size Off. It does come in handy when you have the loop size nailed and are just trying to find the proper start point.



Fix Size Button: When the box is checked, the Loop Size parameter is locked at the current value. This is handy when you have found the right size loop and are looking for a better start point.

Fix Size is very useful for creating groove loops and single cycle loops.

- **Auto Correlate**: Analyzes the waveform to find likely loop points. With **Fix Size On**, the **Start Point** of the Loop is adjusted.

  With **Fix Size Off**, the **End Point** is adjusted.

Using the Fix Size parameter with Auto Correlation

Auto correlation simply means automatic correlation or comparison. The computer analyzes the signal around the loop points you have specified and then moves the start or end point of the loop until it finds a section of the wave that closely matches the section around the start point. If the **Fix Size parameter is On, the Start point** of the loop is adjusted. **If Fix Size is Off, the End point** in the loop is adjusted. Auto correlation may be used again and again with the computer moving the analysis window slightly each time to try to zero in on the optimum loop.

FIR (Phase Linear Filter)

The phase linear filter was originally developed on the EIII to separate the attack and sustain portions of piano samples. Ordinary filters cause serious phase distortion at the cutoff point. This FIR filter has much less phase distortion than a typical filter and so is better suited for sample splicing. The phase linear filter acts on the ENTIRE sample, unlike the parametric filter which can filter any portion of the sample. This ensures strict phase-linearity. There are five types of phase linear filters available: **Lowpass**, **Highpass**, **Bandpass**, **Allpass** and **Notch**.

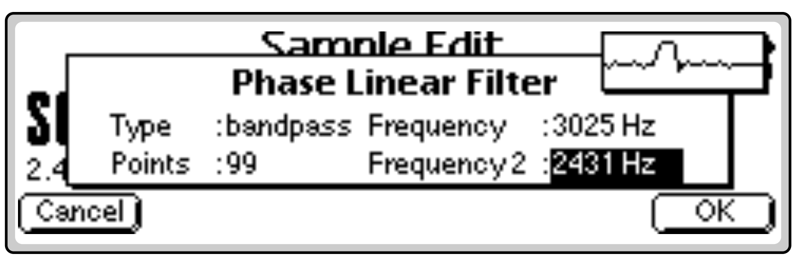
The **Points** parameter determines the Quality or Q of the filter. The larger the Point value, the more the filter behaves like an ideal filter and the steeper the filter slope. The Points parameter is variable from 5 to 99.

The **Frequency** parameter determines the cutoff frequency of the filter and is calibrated in Hertz. When Bandpass or Notch filters are selected, a second frequency field appears to allow you to specify the frequencies on either side of the passband or notch.

The **Allpass** filter is a special purpose filter that is used to add delay to a sample before it is recombined with a filtered sample. Recombining allows you to perform band-splitting or frequency boosting instead of frequency attenuation.

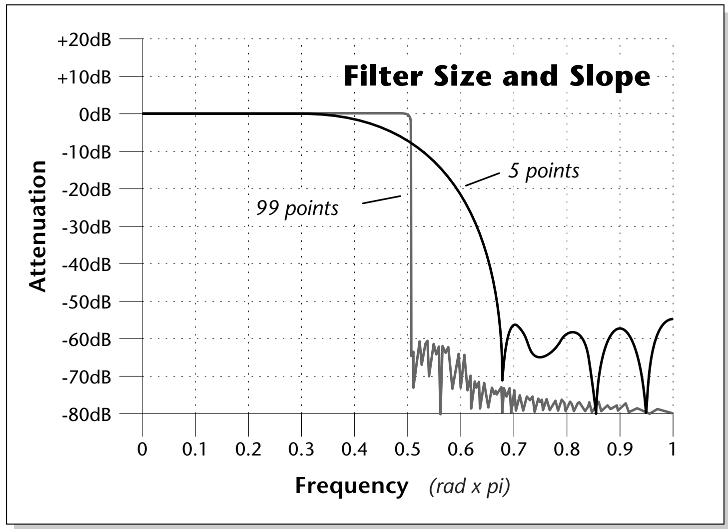
☞ To Filter a Sample:

1. Press the **Sample Edit** key. The LED illuminates and the main sample edit screen appears.
2. **Select the sample** that you want to equalize using the Data Entry Control, INC/DEC keys, or the numeric keypad.
3. Press the **Tools 3** function key (F4). Another row of function keys appears.
4. Press the **FIR** function key (F4). The following screen appears.



5. Select the filter **Type**: Lowpass, Highpass, Bandpass, Notch or Allpass.
6. Set the filter slope by adjusting the **Points** parameter. The higher the number, the steeper the filter slope.
7. Set the **Frequency** parameter to the desired frequency. If you chose the Bandpass or Notch filters, the **Frequency 2** field will appear, allowing you to set the lower frequency in the range.
8. Press **OK** to filter the sample or **Cancel** to cancel the operation.

Holding down the Enter key while turning the Data Entry Control allows "fine tuning" of the value by one number per click.



*This chart shows the relationship between the filter **Points** and **Slope**.*

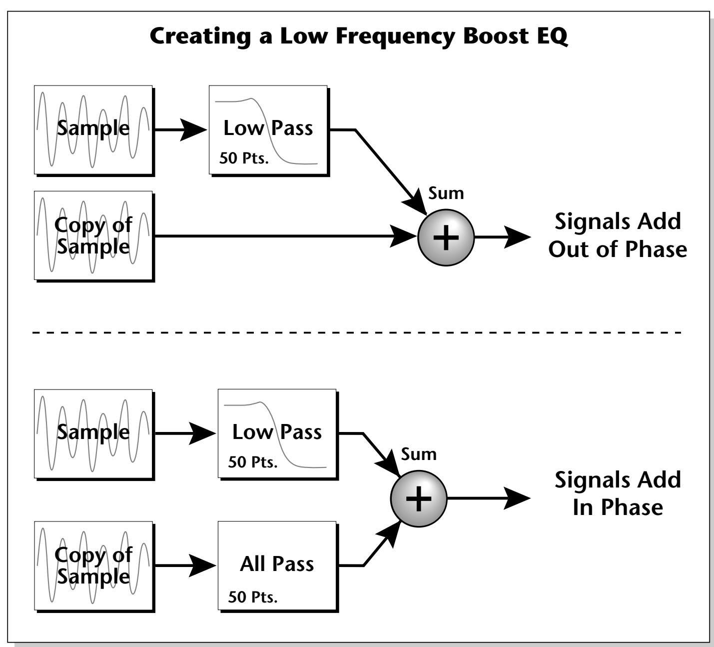
• Alternate Filter Responses

Low Shelving Boost *- Add Lowpass with original signal.*

High Shelving Boost *- Add Highpass with original signal.*

Band Boost *- Add Bandpass with original signal.*

The Allpass filter can be used to achieve complex formants, notches or frequency boosts by *copying*, *filtering* and *adding* the original sample to the filtered version. When adding filtered signals with the FIR, each signal must be processed through the same number of points to preserve phase-linearity. The diagram below shows the incorrect and correct ways to add signals together.



The signal is passed through the Phase-linear filter and recombined with the original signal to create a low frequency boost. Because of the filtering, the two signals are now out of phase. The Allpass filter, set to the same point size, is used to correct the problem.

Bit Converter

For those who want to return to a simpler time when samples were eight bits (or less), we offer the Bit Converter. The Bit Converter takes your pristine 16 bit samples and transforms them into any bit resolution you select. Ever wonder what a 1-bit sample sounds like? Find out with the bit converter. You cannot increase the bit resolution once it's been removed.

Tip #1: *You can create silent samples by setting the bit resolution to zero.*

Tip #2: *The Bit Converter can be used to optimize samples for playback on 8 bit devices.*

☞ To Change the Bit Resolution of a Sample:

1. Press the **Sample Edit** key. The LED illuminates and the main sample edit screen appears.
2. **Select the sample** you want to process using the Data Entry Control, INC/DEC keys, or the numeric keypad.
3. Press the **Tools 4** function key (F5). Another row of function keys appears.
4. Press the **BitCnv** function key (F5). The following screen appears.

5. Enter the desired bit resolution using the Data Entry Control, INC/DEC keys, or the numeric keypad.
6. Press **OK** to process the sample or **Cancel** to cancel the operation.

Preset Edit Menu

Try combining the Pattern LFOs, or controlling the amount of one with another, or combining them with the clock divisors.

LFO Tricks & Tips*:*

- *The Random LFO wave is truly random and is different for each voice and layer.*
- *The Pattern (Pat) waveforms will sound the same on different layers and voices.*
- *Sine + Noise is very useful for simulating trumpet and flute vibrato.*

★ ***When routing Hemi-quaver to Pitch:***

+38 = major scale
-38 = phrygian scale
+76 = whole tone scale
(+38) + (+76) = diminished (two cords)

odd amount = S+H sound

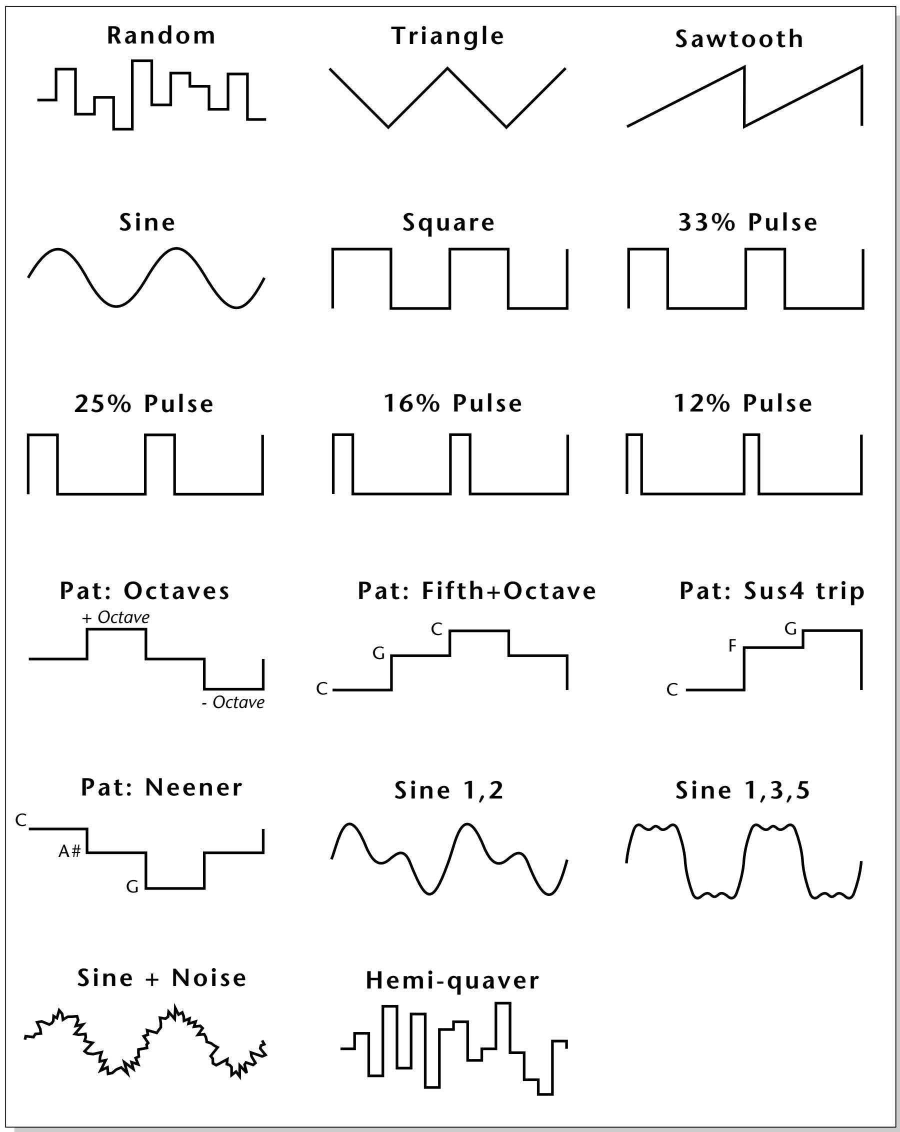
Note: *References to musical intervals in the pattern LFO shapes are with the LFO routed to pitch and a PatchCord amount of +38.*

24 Cords!

EOS now contains 24 PatchCords *(instead of 18)* so you can create more complex patches. *Whoo-hoo!*

New LFO Waveforms

The Emulator has two multi-wave LFOs for each channel and they now have 17 waveforms each! The LFO waveforms are shown in the following illustration.



Disk Menu

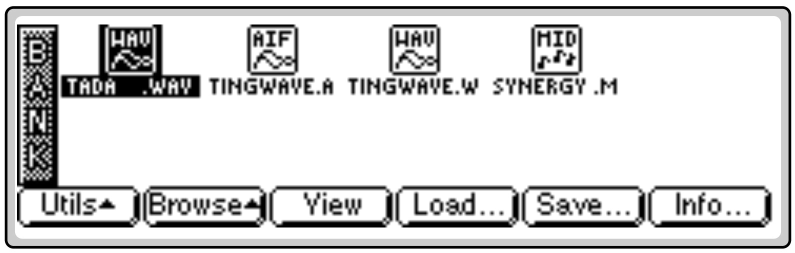
Load .WAV & AIFF files

EOS can load .WAV and AIFF files from a DOS formatted floppy disk. The sample rate is included in the .WAV or AIFF header and are converted to 16 bit samples (filling in zeros if the bit size is less than 16 bits).

Loading samples from floppy can take a LONG time. Unfortunate, but true. It might be a good time to take a break while the sample is loading.

☞ To Load a .WAV or AIFF Sample:

1. Insert a HD DOS formatted floppy containing the sample file.
2. Select **Drives** from the Disk Browser and move the cursor to “Floppy Disk” using the left/right cursor keys.
3. Select **Banks** from the Disk Browser. Each file appears as a Bank containing one Sample. Select the desired **Bank** (sample). The screen below shows a disk containing two WAVE files, an AIFF file and a MIDI sequence.



4. Select **Samples** from the Disk Browser. A single sample icon appears.
5. Press **Load** to load the sample into the current bank.

Sleep

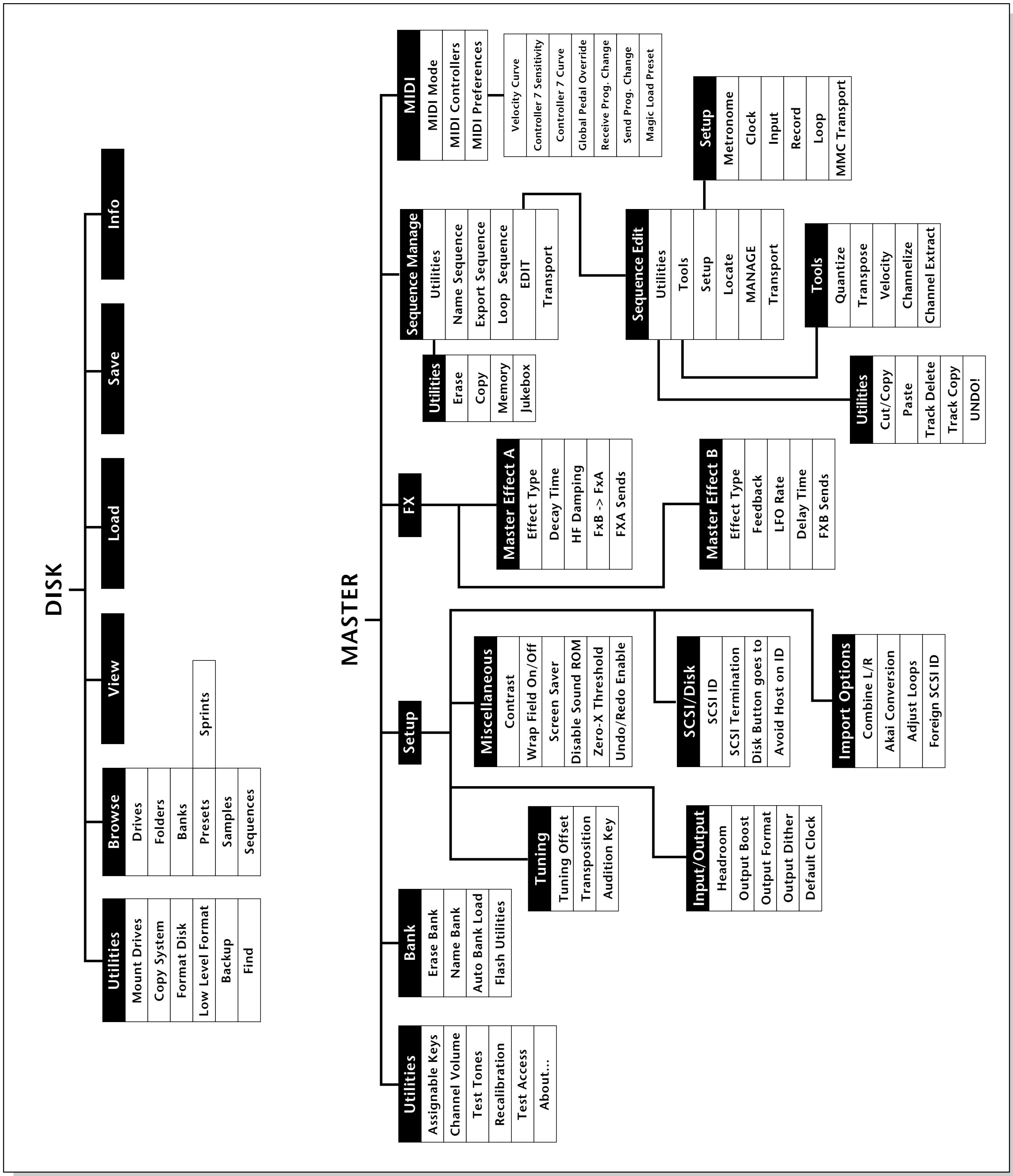
A disk drive can be put to “Sleep” in order to cut down on noise in the quiet studio environment.

Warning: *When chaining multiple Emulators via SCSI, all must be running EOS 4.01 for Drive Sleep to work properly. If you use “Drive Sleep” under these conditions units may freeze, causing data loss.*

☞ To Put a Drive to Sleep:

1. From the Disk Browser, press the **Info…** function key (F6). A pop-up window appears with the vital statistics of the selected disk.
2. To put a drive to sleep, press the **Sleep** function key (F2).
3. The drive will be put to sleep.
4. To wake the Drive, **select the Drive**, then press **Enter**.

Menu Maps


DISK
Utilities
Mount Drives
Copy System
Format Disk
Low Level Format
Backup
Find
Browse
Drives
Folders
Banks
Presets
Samples
Sequences
Sprints
View
Load
Save
Info
MASTER
Utilities
Assignable Keys
Channel Volume
Test Tones
Recalibration
Test Access
About...
Bank
Erase Bank
Name Bank
Auto Bank Load
Flash Utilities
Setup
Tuning
Tuning Offset
Transposition
Audition Key
Input/Output
Headroom
Output Boost
Output Format
Output Dither
Default Clock
Miscellaneous
Contrast
Wrap Field On/Off
Screen Saver
Disable Sound ROM
Zero-X Threshold
Undo/Redo Enable
SCSI/Disk
SCSI ID
SCSI Termination
Disk Button goes to
Avoid Host on ID
Import Options
Combine L/R
Akai Conversion
Adjust Loops
Foreign SCSI ID
FX
Master Effect A
Effect Type
Decay Time
HF Damping
FxB -> FxA
FXA Sends
Master Effect B
Effect Type
Feedback
LFO Rate
Delay Time
FXB Sends
Sequence Manage
Utilities
Name Sequence
Export Sequence
Loop Sequence
EDIT
Transport
Utilities
Erase
Copy
Memory
Jukebox
Sequence Edit
Utilities
Tools
Setup
Locate
MANAGE
Transport
Setup
Metronome
Clock
Input
Record
Loop
MMC Transport
Tools
Quantize
Transpose
Velocity
Channelize
Channel Extract
Utilities
Cut/Copy
Paste
Track Delete
Track Copy
UNDO!
MIDI
MIDI Mode
MIDI Controllers
MIDI Preferences
Velocity Curve
Controller 7 Sensitivity
Controller 7 Curve
Global Pedal Override
Receive Prog. Change
Send Prog. Change
Magic Load Preset

SAMPLE MANAGE

- **Utilities**
  - Erase Sample
  - Copy Sample
  - Sample Dump
  - Defragment Mem.
- **Name Sample**
- **New Sample**
  - Threshold
  - Input - L/R/Both
  - Dither
  - Sample Source
  - Sample Length
  - ADC Gain
  - Arm Sample
  - Force Sample
  - Trigger Key
  - **Automatic**
    - Auto Truncate
    - Auto Normalize
    - Auto Loop
    - Auto Placement
- **Place Sample**
  - Group
  - Original Key
  - Low Key
  - High Key
- **Export Sample**
  - as EIIIx
  - as EIV
  - as WAVE
  - as AIFF
- **Info**

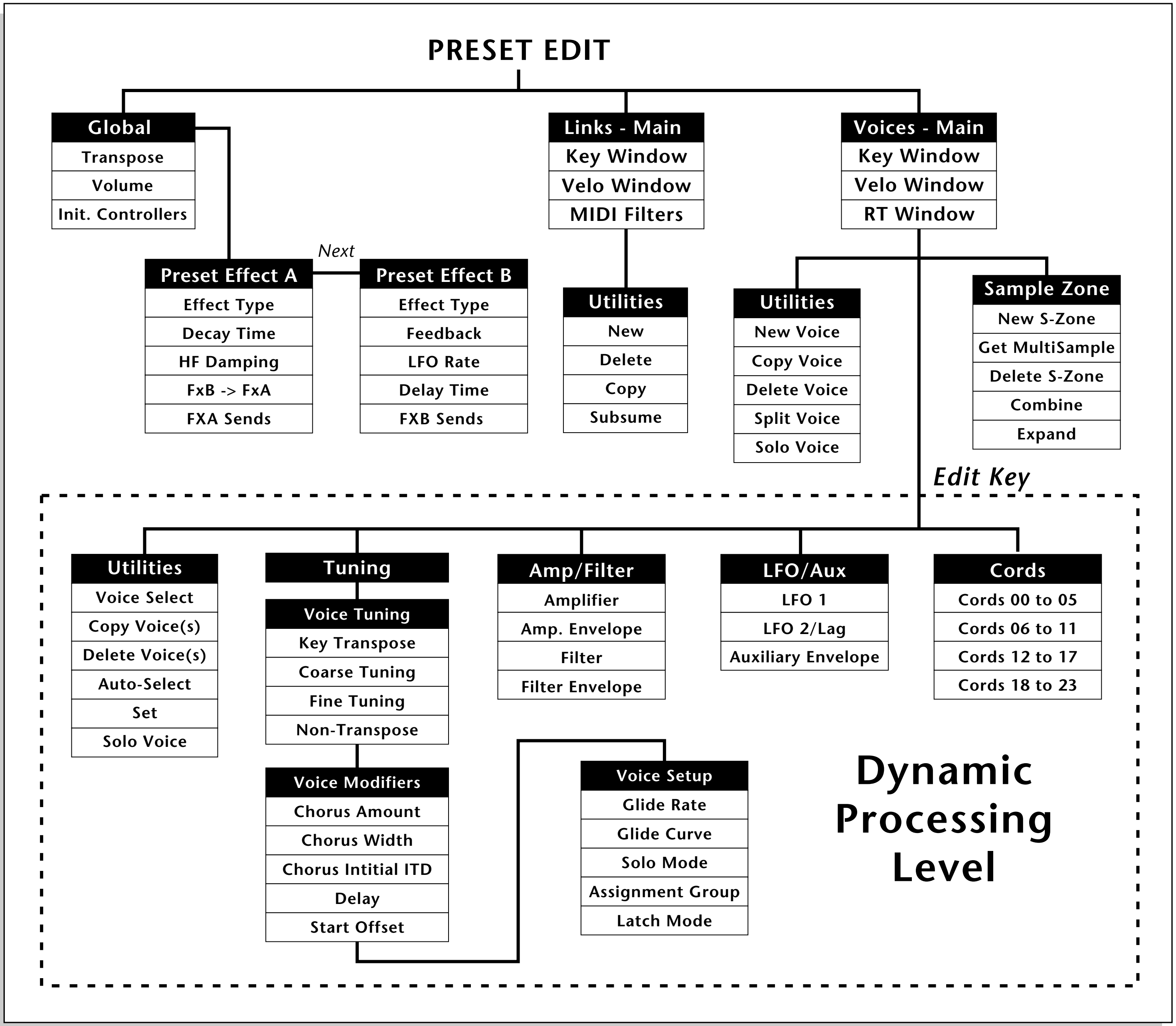
SAMPLE EDIT

- **Utilities**
  - Cut Section
  - Copy Section
  - Paste Section
  - Truncation
  - Taper
- **Tools 1**
  - Loop
  - Loop Type
  - Digital Tuning
  - Sample Rate Convert
  - Sample Calculator
- **Tools 2**
  - DC Filter
  - Swap Left/Right
  - Stereo <-> Mono
  - Reverse Sample
  - Sample Integrity
- **Tools 3**
  - Gain Change
  - Compressor
  - Parametric EQ
  - FIR
  - Aphex Aural Exciter™
- **Tools 4**
  - Transform Multiply
  - Doppler
  - Time Compression
  - Pitch Change
  - Bit Converter
- **UNDO!**

PRESET MANAGE

- **Utilities**
  - Erase Preset
  - Preset Dump
- **Name Preset**
- **New Preset**
- **Copy Preset**
- **Export Preset**
- **Info**

Menu Maps


PRESET EDIT
Global
Transpose
Volume
Init. Controllers
Links - Main
Key Window
Velo Window
MIDI Filters
Voices - Main
Key Window
Velo Window
RT Window
Next
Preset Effect A
Effect Type
Decay Time
HF Damping
FxB -> FxA
FXA Sends
Preset Effect B
Effect Type
Feedback
LFO Rate
Delay Time
FXB Sends
Utilities
New
Delete
Copy
Subsume
Utilities
New Voice
Copy Voice
Delete Voice
Split Voice
Solo Voice
Sample Zone
New S-Zone
Get MultiSample
Delete S-Zone
Combine
Expand
Edit Key
Utilities
Voice Select
Copy Voice(s)
Delete Voice(s)
Auto-Select
Set
Solo Voice
Tuning
Voice Tuning
Key Transpose
Coarse Tuning
Fine Tuning
Non-Transpose
Voice Modifiers
Chorus Amount
Chorus Width
Chorus Intitial ITD
Delay
Start Offset
Amp/Filter
Amplifier
Amp. Envelope
Filter
Filter Envelope
LFO/Aux
LFO 1
LFO 2/Lag
Auxiliary Envelope
Cords
Cords 00 to 05
Cords 06 to 11
Cords 12 to 17
Cords 18 to 23
Voice Setup
Glide Rate
Glide Curve
Solo Mode
Assignment Group
Latch Mode
Dynamic
Processing
Level

World Headquarters
E-mu Systems, Inc.

P.O. Box 660015
Scotts Valley, CA USA
95067-0015
Telephone: 831-438-1921
FAX: 831-438-8612

Europe, Africa, Middle East
E-mu Systems,ltd.

Suite 6, Adam Ferguson House
Eskmills Industrial park
Musselburgh, East Lothian
Scotland, EH21 7PQ
Telephone: +44 (0) 131-653-6556
FAX: +44 (0) 131-665-0473