

Emulator[®]X3

STREAMING SAMPLING SYNTHESIZER

Reference Manual

 **E-MU**



Reference Manual

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Version 3.0

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Table of Contents

1 - Introduction & Installation	15
Introduction	15
Emulator X3 Features	15
Streaming	15
Sampling	15
Synthesizer	16
Before you Begin... ..	16
Notes, Tips and Warnings	16
Tutorial List	16
What's New in X2	17
New in X3	17
Minimum System Requirements	18
Software Installation.....	18
Programs installed:	18
Running the Application for the First Time.....	19
Setting up your Audio.....	20
Setting up the Preferences	21
Audio Preferences	22
Streaming	22
Audio Setup	23
MIDI	26
MIDI Inputs	26
MIDI Response	27
Controllers	28
Appearance Settings in Windows	29
Other Settings	29
2 - Emulator Architecture	31
Organization of the Emulator X Modules	31
A Modular System	32
Bank	32
Saving	32
Undo/Redo	32
Multisetup	34
Preset	34
Voice	34
Sample	35
Flexible Architecture	35
Preset Diagram	36
Voices	36
Multisample Voices	37
The Sample	38
TwistaLoop™	39

3 - Getting Started 41

Exploring the Tree	41
Multisetup Page.....	42
Changing Presets (from Single View)	43
Multi Channel View	44
Changing Presets (from the Multichannel view)	45
Adjusting the Controllers	45
Create Your Own Custom Bank	47
Load a New Bank	49
Opening a Preset	49
Examine a Voice	52
Wipe the Slate	53
Examine the Samples.....	54
Using DSP	55
Experiment!	57
Save and Save As	57
Emulator X Bank Structure	57
Building a Preset from the Ground Up.....	58
Emulator X Advanced Applications Guide	65
The Librarian	66

4 - Multisetup / Preset Globals 69

The Multisetup.....	69
Changing Presets (from Single View)	70
Multi Channel View	71
The Current MIDI Channel	72
MIDI Channel	72
Preset Assignment	72
Channel Volume	72
Channel Pan	72
Output Assignment	72
Output View Button	73
Aux FX View Buttons	73
Aux Send On/Off	73
Preset FX Enables	73
Select Effect	73
TwistaLoop Override	73
Global Controls.....	74
Master Volume Control & VU Meter	74
Tempo Control	74
Master Tune	74
Master Transpose	74
Limiter	74
Aux Outputs	75
Using the Emulator Sessions in PatchMix DSP	75
Filter Override	77
MIDI Controllers & Scribble Strips	77
Pre-roll RAM Used	77
CPU Meter	77
Number of Samples Playing	77
Current Channel	77
Working with Multisetups.....	78
Saving FX & Multisetups	78
VSTi Multisetups	78

Preset Globals.....	80
Transpose & Volume	81
Initial Controllers A-P	81
Capture MIDI Controllers	82
TwistaLoop Lock To Tempo	82
Initial Tempo	82
Tuning Tables	83
Key CC Window	85
Preset Modulators	85
Channel Lag 1 & 2 Rate	85
Poly Key Timer	86
Channel Ramp	86
Preset PatchCords	87
Templates	87

5 - Voices & Sample Zones 89

Voices and Zones Pages.....	90
Mix/Tune	90
Key Win	90
Vel Win	90
RT Win	90
CC Win 1-5	90
Region	90
Details	90
Voice Selection	91
Manipulating Voices	91
Working With Multiple Voices	91
Samples & Multisamples	92
Groups	93
Sorting Voices and Zones	93
The Keyboard Display	94
Overview Section.....	95
Sample Controls	95
Sample	95
Original Key	95
Group	95
Direction	95
Mix Tune Controls	96
Mute	96
Solo	96
Volume	96
Pan	96
Fine Tune	96
Coarse Tune	96
Transpose	96
Key Win, Velocity Win, RT Win, and CC Win Controls	96
Direction	97
Adding Samples and Voices.....	97
Mix/Tune Page.....	99
Key Window Page	99
Using Values to Define Key Window Parameters	102
Velocity Window Page.....	103
Using Values to Define Velocity Window Parameters	104
Realtime Window Page.....	105

Using Values to Define Realtime Window Parameters	107
Continuous Controller Window Pages.....	108
Keyboard CC Window.....	109
Cycle Groups	111
Using Values to Define CC Window Parameters	114
Region Page	115
Creating or Editing a Region	116
Region View Controls	116
Sample Zoom and Scroll Controls	116
Sensitivity	116
Snap Settings and Markers	117
Region Xplode	117
More About Regions	118
Details Page	119
Voice & Sample Zone Utilities	120
Open Voices	120
Open Sample	120
New Voice	120
Delete Voices	120
Duplicate Voices	120
Combine Voices	120
Expand Voice	121
New Sample Zone	121
Delete Sample Zones	121
Duplicate Sample Zones	121
Sort Voices	121
Sort Zones	121
New Group	121
Select Group	121
Clear All Solos and Mutes	121
Edit Menu Utilities - Cut, Copy & Paste.....	122
Links.....	123
Link Utilities	124
Subsuming Links	124
Mix/Tune Window	125
Key Window	125
Velocity Window	125
CC Window	126

6 - Voice Processing 127

Simplified Emulator X Signal Path.....	127
Overview of the Voice Processing Window.....	128
Templates	128
Oscillator.....	129
Pitch Bend Range	129
TwistaLoop Controls	129
Speed	129
Loop (% or number)	129
Start At Loop	129
Tuning Controls	129
Transpose	129
Coarse Tuning	129
Fine Tuning	130
Fixed Pitch Mode	130

Chorus Amount	130
Tips for Using Chorus	130
Glide Rate & Curve	131
Key Controls	131
Delay	131
Sample Offset	131
Sample Start Offset	131
Amplifier, Filter & Auxiliary Envelopes.....	132
Adjusting the Envelopes	132
Envelope Mode: Repeat	133
Envelope Mode: Complete	134
Time-based and BPM Envelopes	134
BPM Envelopes	134
BPM Tempo Chart	135
LFOs & Lags	136
Frequency	136
Delay	136
Shape	136
Variation	136
Sync	137
BPM	137
LFO Tricks & Tips:	138
Lag Processors 1 & 2	138
Extra Summing Amps	138
Function Generators	139
Function Display and Grid	139
Function Generator Main Controls	141
Modulation Destinations	143
Function Generator Ideas	143
Filter Controls	145
Filter Descriptions	145
Programmable Morphing Filters	149
Morph Designer	149
Dual EQ Morph	151
Dual EQ Morph + Expression	152
2EQ + Lowpass Morph	153
Peak/Shelf Morph	153
Amplifier Controls	154
Volume	154
Panning	154
Amp Envelope Dynamic Range	154
Response	154
FX Wet/Dry	154
Voice Effect Sends	155
Modulation Cords.....	156
Pitch Amount Chart	158
Modulation Source Polarity	160
Summing Nodes	161
PatchCord Lore	161
Sample Loop Modulation	161
Assign Groups	164
Output Assign Groups	164
Poly Key Assign Groups	165
Keyboard Modes	166
Latch Mode	167

7 - Editing a Sample 169

Sample Editor Window	169
Exporting & Importing Samples	171
Exporting a Tempo Map	171
Tips for Exporting Tempo Maps	172
Importing a Tempo Track to a Sequencer	173
Display Menu Items.....	174
Zoom In (Page Up)	174
Zoom Out (Page Down)	174
Zoom In (Ctrl +Page Up)	174
Zoom Out (Ctrl +Page Down)	174
Zoom To Selection (F10)	174
Zoom To All (F11)	174
Zoom To Sample (F12)	174
Display Time in Samples	174
Display Time in Seconds	174
Display Time in Bar:Beats	174
Display Level in Percent	174
Display Level in Decimal	174
Display Level in dB	174
Show Note Markers	175
Show Loop Markers	175
Show Zero Crossing Threshold	175
Vertical Units	175
Horizontal Units	176
Transport/Loop Controls	177
Play (from Cursor)	177
Play from Start	177
Stop	177
Loop Play	177
Go to Sample Start	177
Go to Sample End	177
Go to Loop Start	177
Go to Loop End	177
Move Start to Previous Zero	177
Move Start to Next Zero	178
Move End to Previous Zero	178
Move End to Previous/Next Zero	178
Autocorrelate Loop	178
Cut, Copy, Paste & Delete	179
Undo and Redo	179
Typical Applications	179
Cut Section	180
Copy Section	180
Paste Section	181
Sample Edit Tools	185
MIDI Audition	185
Truncate	186
Fade	186
Normalize	187
Adjust Gain	187
DC Filter	188
Reverse	189
Swap L-R Channels	189

Bit Reduction	190
Time Compression	190
Pitch Shift	192
Transform Multiply	193
Convert to Mono or Stereo	194
Sample Rate Convert	195
Xtractor	196
Vocal or Instrument Isolation/Elimination	198
A Visual Mix Analysis	200
Batch Processing.....	202
Looping and Loop Processing	203
Crossfade Looping	204
Truncate after Loop	204
Loop Compression	204
Loop Settings	205
Loop Button	205
Loop in Release Button	205
Loop Select	205
Lock Button	205
Select Loop	205
Select All	205
No Snap	205
Snap To Marker	205
Snap To Zero-Crossings	206
Background: Zero Crossing Threshold	206
Delete Unused Samples	206
How to Loop a Sample	207
Zero Crossing	208

8 - TwistaLoop **209**

Introduction	209
Analyzing Beats In the Sample Edit Window.....	209
Running Beat Analysis	209
Tempo Range (BPM)	210
Tempo Variation	210
Beats per Bar	210
Meter	211
Create note markers	211
Create beat markers	211
Viewing Your Analyzed Sample	211
Markers	212
The Triplet Button	212
Sensitivity	212
Fine-Tuning Your Analyzed Sample	213
Shifting the Beats	213
Beat Parameters	213
Bar Numbering Changes	214
Beats Per Bar Changes	214
Tempo Granularity Changes	214
Creating Loops from Your Analyzed Sample	215
Tips for Making Selections for Loops	216
Using Loops.....	217
Playing Back Loops	217
Loop Speed and Tempo	219

Changing Loops Using a Continuous Controller	219
Loop Select (Continuous)	220
Loop Select (Jump)	221
Other Ways to Change Loops	222
Synchronizing Loops	222
Multisample Voices and Looping	223
The Speed Knob	224
Changing a Loop's Speed	224
Nudging the Tempo	224
Using Regions of Analyzed Samples	226
Using the RT Win With Regions	227
Troubleshooting.....	228
Beat Analysis Doesn't Find the Right Beats	228
Looping Doesn't Occur	228
Loops Don't Synchronize to Tempo	229

9 - Effects 231

Overview	231
Programming Effects.....	232
Adding an Aux Effect in the Multisetup	232
Adding an Effect to a Preset	233
Block Diagram of the Effects Routing	234
Block Diagram of the Effects Screens	235
Effects Overhead	236
Typical Effects Usage	236
ASIO Outputs	236
Main Output Window	237
Main Output Window	237
Proteus X Composer FX Template.....	238
Key Points	238
Voice	238
Preset	238
Multisetup	238
Controlling the Preset Effect Amount from the Voice	239
Aux FX Screen	240
Aux Effects	240
Aux FX Send Amounts	240
Preset FX Screen.....	241
Preset Effects	241
Preset FX Modulation Parameters	242
Creating and Deleting Templates	242
Effects Listing.....	243
Effects Descriptions.....	243
Background: Reverb	243
Reverb	244
Early Reflections Reverb	244
Reverb Lite (mono)	245
Chorus	246
Background: Chorus	246
Chorus / Delay	247
Compressor	248
Basic Controls	248
Delay	250
Background: Delay	250

Discrete stereo	250
Ping Pong, stereo	251
Ping Pong, mono L->R	251
Delay (BPM)	253
Early Reflections	254
1-Band Para EQ	254
4-Band EQ	255
Flanger	256
Flanger (BPM)	257
Growl	258
Limiter	259
Phaser	260
Pitch Shifter (mono)	261
Ring Modulator	262
SP12-ulator	263
Tremulator BPM	264
Tube	266
Twin	267

10 - Controls 269

The Toolbars.....	269
Viewing and Hiding the Toolbars	270
Drag & Drop	270
Floating Keyboard.....	270
Changing Settings	271
Entering Numeric Values	271
Adjusting Ranges & Fades.....	272
Selecting Voices using your MIDI Keyboard	273
Adjusting Envelopes.....	274
BPM Mode & Tempo	274
Cut, Copy & Paste	274
View Menu	275
List Display View	276
Creating a Text List of Presets and Samples	276
Tree View	277
Tree Navigation using the Keyboard	277
Cursor Keys	277
Find Items	278
Tree Navigation Using the Toolbar	278
Refresh (F5)	278
Context Sensitive Menus	278
All Sound Off (Ctrl + Pause)	278
Templates.....	279
Creating and Recalling Templates	279
Organizing Templates	280

11 - From Sample to Preset 281

Acquire (Sampling) 281
 Input Controls 281
Recording Audio (Sampling) 282
 Record Audio Controls 283
Gating the Samples 284
 Gating Display 284
 Gate/Chop Controls 284
 Region List Display 285
 Region List Controls 286
Saving your Samples 288
 Save to Sampler Controls 289
SynthSwipe™ 290
 Tips for Saving SynthSwipe Recordings 293
Resampling 293
Importing Other Sample Formats 294
 Set Original Keys 295
 Original Key Algorithms 295
Sample Options 296
Build Preset Options 297

12 - Synthesizer Basics 299

Emulator X Advanced Applications Guide 299
Editing Presets 299
Modulation 299
 Modulation Sources 300
 Keyboard Key 300
 Key Velocity 300
 Release Velocity 300
 Gate 300
 Key Glide 300
 Pitch and Mod Wheels 300
 Keyboard Pressure (mono aftertouch) 300
 Pedal 300
 Miscellaneous Controllers A-P 301
 Low Frequency Oscillators (2 per voice) 301
 Envelope Generators (3 per voice) 301
 Function Generators (3 per voice) 301
 Noise & Random Generators 301
 T-switch and Footswitches 301
 Modulation Cords 301
Envelope Generators 302
 Retriggering 302
 Complete Mode 302
Low Frequency Oscillators (LFOs) 303
Random Sources 304
Clock Modulation 304
 Syncing an LFO to the Clock 304
Function Generators 305
Modulation Destinations 306
Modulation Processors 307
 Modulation Processor Examples 308
Dynamic Filters 309

What is a Filter?	310
Parametric Filters	312
The Z-Plane Filter	313
MIDI Channels & Real-time Controls.....	314
Assigning MIDI Controller Knobs.....	316
MIDI Program Change Commands	317
MIDI Bank Select Commands	317
Background: Bank Select Commands	317
MIDI Modes.....	317

13 - Appendix 319

Translator Emulator X Converter	319
Supported File Types	319
Emulator X VSTi	320
Launching the VSTi Application	320
1 - Setting up Cubase LE4	320
2- Recording a MIDI Track using Emulator X3	322
On Your Own	325
Multisetup is Saved in the Recording Application	325
Checklist for Launching Emulator X VSTi	326
Cubase LE	326
If you're using PatchMix DSP...	326
Keyboard Shortcuts.....	327
Drag & Drop	328
Toolbar Functions.....	331
MIDI Key Number to Note Chart.....	332
Frequency Range of Common Instruments	333
MIDI Implementation Chart	334
Received Channel Commands	339
Special Notes:	339
Optimizing Your Computer for Emulator X	340
SCSI and IDE Drive Tips and Considerations	340

Index 341

1 - Introduction & Installation

Introduction

Congratulations on your purchase of the Emulator X3 Streaming Sampling Synthesizer—the most powerful and complete software instrument in the world, built on E-MU's 35 years of synthesis experience. The Emulator X3 features the Xstream™ streaming sound engine with up to 192kHz sampling, and playback, an arsenal of automated sampling/beat analysis/editing tools and almost unlimited sound manipulation.

Emulator X3 Features

Streaming

- Xstream™ 24-bit/192 kHz Streaming Engine, featuring E-MU's patented pitch interpolation and precision 32-bit floating point processing for unmatched sound quality.
- 32-bit and native 64-bit applications included (32-bit: Windows XP and Vista; 64-bit: Windows x64 and Vista)
- Optimized for Multi-core and Multi-Threaded processors to intelligently balance the processing load across all available resources for exceptional performance.
- Standalone (64 MIDI channels) or VSTi (16 MIDI channels per instantiation) operation.

Sampling

ASAP™ Automated Sampling and Placement

- Sample, Chop and Place
- Load, Chop and Place
- SynthSwipe™ automated hardware and software instrument sampling

Integrated Sample Editor

- Auto-Correlation Looping
- Crossfade Looping
- Comprehensive DSP Tools including Transform Multiply
- External Editor Link

Beat/Loop Tools

- Beat Analysis and Markup
- TwistaLoop™ non-destructive audio manipulator
- Xplode™ beat slicer with MIDI and Tempo Map Export

Synthesizer

- Advanced Modular Synthesis and FX Engine (from your initial sample's tempo through Emulator X3's entire synthesis and FX architecture, all time-based parameters syncable to BPM clock).
- Over 50 types of Z-Plane™ Morphing Filters (2nd order to 12th order), including the Morph Filter Designer
- Multi-Function Generators/Arpeggiators (can be programmed to modulate pitch, filters, volume, retrigger samples, or LFOs, change tempo or trigger other events)
- Virtual PatchCord Modulation Matrix (over 100 modules)
- Integrated Modulation FX engine (23 core effects), including SP12ulator, Tremulator, Reverbs, Delays and many more
- Comprehensive sound format support, including EOS, EIII, GigaSampler, MP3, REX2, .AIFF, SoundFont 2.1, .WAV and more.

Before you Begin...

You should have a good working knowledge of your computer's operating system. For example, you should know how to use the mouse, standard menus and commands. You should also know how to cut, copy, paste, open and close files.

Notes, Tips and Warnings

Items of special interest are presented in this document as notes, tips and warnings.

- ◆ **Notes** provide additional information related to the topic being discussed. Often, notes describe the interaction between the topic and some other aspect of the system.
- **Tips** describe applications for the topic under discussion.
- ▼ **Warnings** are especially important, since they help you avoid activities that can cause damage to your files, your computer or yourself.

Tutorial List

Although this document is primarily intended as a reference manual, it does contain quite a few step-by-step tutorials that you may find helpful. A few are listed below.

- **Tip:** Please refer to the "[Index](#)" for a complete list of all Emulator X tutorials.

Tutorial	Page	Tutorial	Page
Adding an Aux Effect	232	Function Generator	140
Adding an Effect to a Preset	233	Creating Sequencer Tempo Tracks	173
Building a Preset from Scratch	58	Layering Presets	124
Change Loops with a Controller	220	Looping	207
Conserving Polyphony	165	Morph Designer	150
Control FX Parameters via MIDI	242	Sampling	282
Create a Custom Bank	47	Program Change Commands	317
Creating ADSR Envelopes	133	Setting up your Controller Knobs	28
Crossfade between two Loops	227	SynthSwipe	290
Examine a voice or sample	52	VSTi	320

What's New in X2

- Support for 64 MIDI channels ([page 70](#))
- Three 64-stage Function Generators per voice ([page 139](#))
- Morph Designer allows you to design custom morphing filters ([page 149](#))
- TwistaLoop loop analysis and manipulation tools ([page 209](#))
- User interface changes
 - New Region Page ([page 115](#))
 - New Detail Page with Voices and Zones info ([page 119](#))
 - Improved Keyboard Display ([page 99](#))
 - Floating Keyboard ([page 270](#))
 - Tabs in Voice Editor ([page 89](#))
 - Envelope Mode Settings ([page 132](#))
 - Sample Display/Overview ([page 169](#))
 - Sample Root Note shown with Red Bar ([page 99](#))
- Key CC Window Controls ([page 108](#))
- Cycle Groups in CC Window provide more natural sound by cycling through different samples on repeated key presses. ([page 111](#))
- Poly Key Groups allow you to limit polyphony per key ([page 164](#))
- New modulation sources for patch cords: ([page 156](#))
 - Function Generator (+,~,<,)
 - Function Generator (gate, trigger)
- New modulation destinations for patch cords: ([page 159](#))
 - Function Generator Rates/Retrigger/Length/Direction
 - Key Timer Rate
 - Lag Rates 1 & 2
 - Loop Select: Continuous/Jump
 - Sample Retrigger (+, -)
 - TwistaLoop Oscillator Speed
- New Effects
 - Compressor ([page 248](#))
 - Limiter ([page 259](#))
 - Tremulator BPM ([page 264](#))
 - Flanger BPM ([page 257](#))
- New Tuning Tables: al-Farabi 12, 22-Tone, 24-Tone, 36-Tone ([page 83](#))
- More templates (XML based)
- Capture MIDI Controllers to preset initial settings ([page 82](#))
- Voices and Zones can be sorted according to different criteria ([page 121](#))
- New Preference Settings: Select edited preset for multisetup, Expand tree items when selected, Knob tracking ([page 29](#))
- New MIDI Setting: Preset Tempo Channel ([page 27](#))
- More Undo Levels and greater control over undo behavior ([page 30](#))
- Effects Overhead dialog box ([page 236](#))

New in X3

- Xstream™ 24-bit/129kHz Streaming Engine—32-bit and native 64-bit applications optimized for Multi-core and Multi-threaded processors. Vista compatible.
- Xplode Region can now create companion MIDI files ([page 117](#))
- Export Tempo Map from Beat Analyzed samples ([page 171](#))
- Xtractor tool selectively processes audio based on stereo position ([page 196](#))
- Librarian improvements - Faster operation with large libraries ([page 66](#))

Minimum System Requirements

To use Emulator X you will need at least:

- Pentium IV 1.5 GHz or equivalent CPU (2.4 GHz or faster recommended)
- CD-ROM Drive
- 512 MB RAM (2 GB DDR or greater recommended)
- Windows XP, Windows x64, Windows Vista, or Windows Vista x64
- Onboard Sound or Audio Interface

Software Installation

Programs installed:

- Emulator X3 Streaming Sampling Synthesizer application
- Emulator X VSTi
- Emulator X Factory Sound Banks

◆ **No Hardware Requirement:** As of software version 3.0 Emulator X3 no longer requires that a piece of E-MU hardware be connected to your computer system.

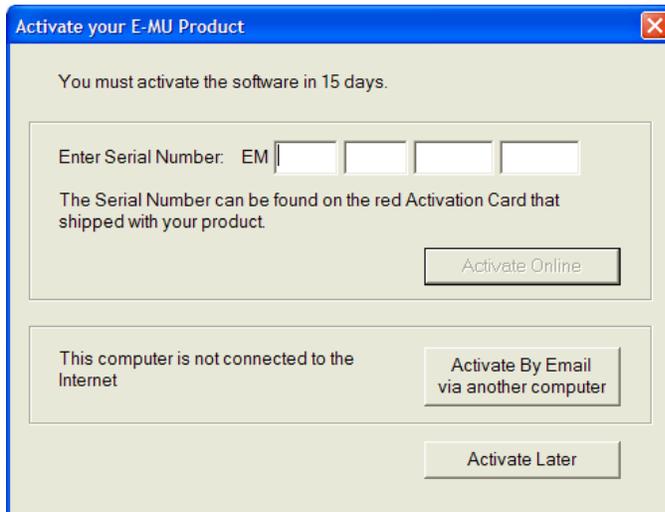
1. Insert the CD labeled "Emulator X3 - Software/Manual/Sounds Installation CD" into your CD-ROM drive.

If Windows AutoPlay mode is enabled for your CD-ROM drive, the CD starts running automatically. If not, navigate to the CD and double-click on **Setup.exe**.

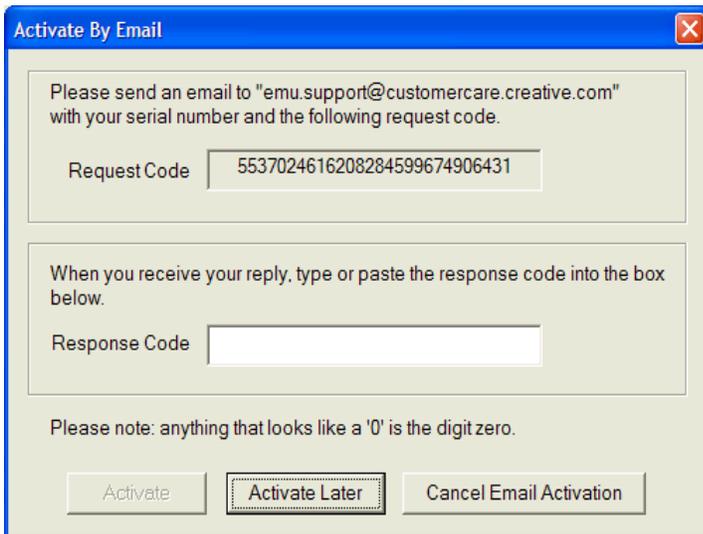
2. Follow the instructions on the screen.
3. You will be asked to choose the Destination Folder for the program. The default location is: "Program Files\Creative Professional\Emulator X."
4. You will be asked to choose the Destination Folder for the VST plugins. The default location is: "Program File\Steinberg\VstPlugins"
5. The program will be installed. Click **Finish** to complete the installation (when prompted) then remove the Emulator X application disk from the CD drive.
6. The included CDs contain the following Emulator X sound banks:
 - X Producer.....E-MU General MIDI, Hip Hop Producer, Proteus X Composer, Saint Thomas Strings
 - Studio Grand..... A stunning 1.4 GB grand piano bank
 - Beat Shop 1ATC, Buju, Head Nod, Homage, IDA, Modern Kit, Rock Kit, Seattle, Studio Kit
 - Xperience.....Breakdown, Filters and Functions, Guitar Wah Bazaar, Jam Session, Jazz Drummer, Transform Tool Kit, Transformed, Twista Mania
 - Xtreme Lead-1...The complete set of presents and samples from the Proteus Xtreme Lead-1 sound module
7. Follow the onscreen instructions to install the banks on your hard disk. The Factory Sound Banks are installed inside "Program Files\Creative Professional\Emulator X\Production Sound Banks" by default.
8. You may install the sound bank disks now or at a later time.

Running the Application for the First Time

1. Launch the Emulator X program by double clicking the Emulator X icon on the desktop or by selecting the Emulator X item from the Start menu.
2. You will be asked to activate your copy of the software by entering the serial number included with your master CD-ROM.



3. Activation can be verified via the internet or by email. If your computer is connected to the internet, activation will be automatic.
4. If you are NOT connected to the internet, you can register via email using another computer (such as a friend's house, or public library). Click the "Activate by email via another computer" button. The following dialog box appears.

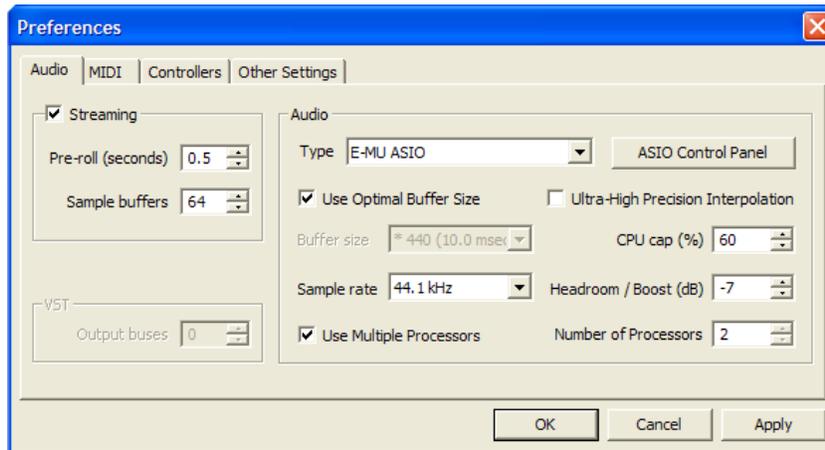


5. Contact E-MU by email at: emu.support@customercare.creative.com. Be sure to include your **Serial Number** (from your Activation Card) and the **Request Code** in the email message. You will receive a return email with your authorization code.
6. When you receive the response code enter it in the box above and click "Activate".
7. Continue on to "Setting Up Your Audio" on the following page.

Setting up your Audio

Before you can start making music, you have to establish a connection between Emulator X and your audio output hardware. This is done by selecting a software driver that makes this connection between hardware and software.

1. In the **Options** menu of Emulator X, select **Preferences**. The following screen appears.



2. In the Audio section of the menu, select **E-MU ASIO** if using an E-MU Digital Audio System. Select **ASIO E-MU USB|0202**, **ASIO USB|0404** or **ASIO E-MU Tracker Pre|USB** if using an E-MU USB interface. If you're using another brand of audio interface, select the appropriate driver.
3. You can verify that the Emulator X is producing audio by clicking on the  icon in the toolbar. This calls up the **Floating Keyboard**. Play the floating keyboard with your mouse and you should be hearing sound and seeing activity on the Emulator X main mix meters.
4. After running Emulator X, you may want to adjust the Audio Preferences for optimum performance on your computer system.
5. If for some reason you want to use hardware that doesn't support ASIO, select the **DirectSound** driver. For Emulator X to communicate with your sound card, you need to have Microsoft DirectSound installed on your computer and there must be a DirectSound driver installed for your audio hardware.

Please go to [page 22](#) for complete explanations of the Audio Preferences.

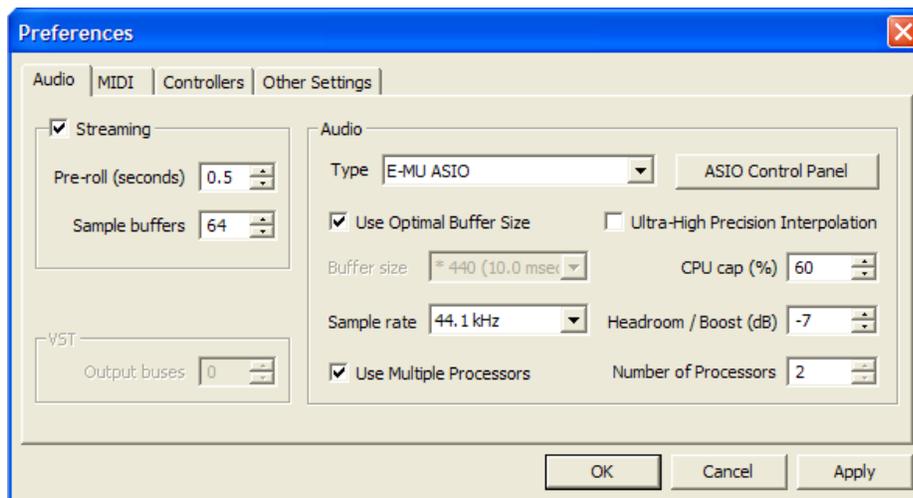
Setting up the Preferences

The preferences dialog box contains a variety of important controls used to configure Emulator X for your audio hardware and to customize the controls to suit the way you work.

To get the best possible performance from your computer, you'll need to spend a little time adjusting the Audio Preferences such as Pre-roll, Sample Buffers, CPU Cap and ASIO Buffer latency. These parameters interact with each other and you may have to experiment to find the settings that work best on your unique computer system.

► To Access the Preferences Dialog Box

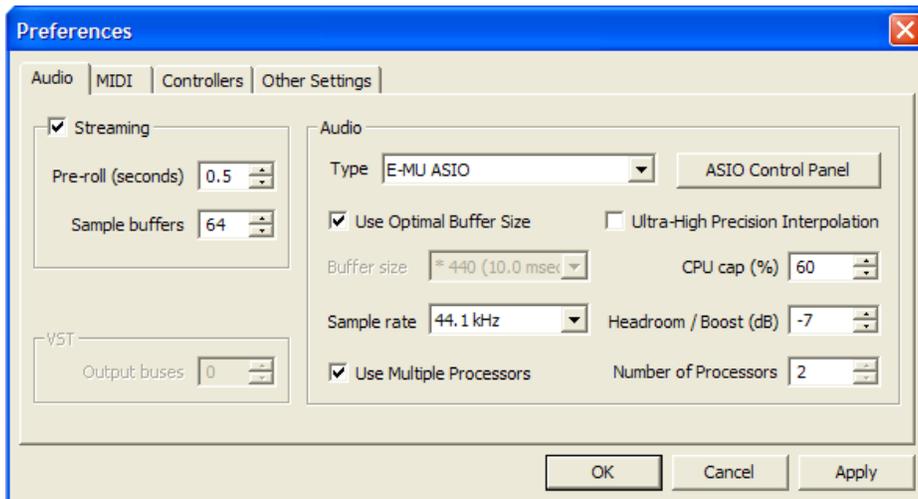
1. In the Options menu, select Preferences. The following screen appears.



2. The preferences dialog box is divided into four main groups. Click on the tab heading to select one of the four sets of controls.

Audio	Disk Streaming Controls, Sample Rate, Audio Hardware Setup & Pitch Shift Algorithm Selection, Multiple Processor Support, ASIO Control Panel.
MIDI	MIDI Input Channels 1-64, IntelliEdit, and Receive/Ignore Program Changes, Internal/External Tempo Source select, Velocity Curve, Volume Curve, Volume Sensitivity, Preset Tempo Channel.
Controllers	This section allows you to set up the MIDI Continuous Controllers you want the Emulator X to receive.
Other	This group contains several miscellaneous controls: Load last bank at startup, Select preset on active channel when editing in multimode, Clear clip indicators after 10 seconds, Expand tree items when selected, Knob tracking, Recording Directory, External Audio Editor, Undo settings.

Audio Preferences



Streaming

When you load an Emulator X bank (with Streaming on), you are not actually loading the sample files themselves into your computer's RAM. Only the preset information and the first few seconds of each sample are loaded. The rest of the sample data is taken directly from the hard disk when needed.

This streaming technique makes fast loading times possible even when using huge banks that would clog a RAM-based sampler. Multi-layer, minute-long, stereo samples on each key are no problem for Emulator X. Sample streaming technology is not new. It was originally developed in the 1980's for hard disk audio recorders so that fast punch-ins and punch-outs could be achieved.

You have the option to enable or disable streaming audio. If streaming is turned off, Emulator X loads (or attempts to load) the entire bank into RAM.

If streaming is so great, why would you ever want to turn it off? Well, if you're using your computer as an audio workstation, performing multiple simultaneous functions, sooner or later, you'll exceed your computer's CPU, memory or disk access resources. Sample streaming makes very heavy demands on your hard disk. If you're recording other audio tracks, using multiple software plug-ins and playing Emulator X with a lot of voices, you may run into a disk access bottleneck. By disabling streaming and loading the bank into your unneeded RAM, your hard disk can be reserved for more critical operations.

Pre-roll (seconds)

This important control allows you to set how many seconds of each sample should be loaded into RAM when the bank is loaded. This setting greatly influences the performance of Emulator X when streaming is on. The setting you choose depends on: the number of samples you want to use and the amount of RAM you want to dedicate to Emulator X.

The smaller the pre-roll, the more disk access will be required for sustained sounds. Frequent disk access may overtax your computer system, depending on its speed and the number of other disk dependent applications you have running. Large pre-roll times improve polyphony and performance at the expense of system RAM and slower bank loading. If you set the pre-roll high enough, the entire bank will be loaded into RAM and no disk streaming will occur.

■ **Tip:** If you get an error message when trying to open a file, the file may be too large to fit into RAM. Turn streaming on.

■ **Tip:** Decrease Pre-roll time for fast load times when auditioning sounds and creating new banks.

Increase pre-roll time when you're playing many notes or dense sequences.

Sample Buffers

This control allocates the amount of RAM that will be set aside for Emulator X (in other words, the number of the largest complete samples in the bank that can be kept in the RAM cache). As samples are played, they are streamed from the hard disk into RAM. If you play those samples again and they are already in RAM, they don't have to be reloaded. This control sets the size of the RAM cache that will be kept and lowers disk access at the expense of RAM.

■ **Tip:** Adding RAM is a relatively cheap and easy way to improve EX3 performance. With more RAM available, you can turn up the Pre-roll and Sample Buffers to reduce disk access.

► Optimize the Performance and Polyphony of your System

The polyphony of the Emulator X is dependent on a variety of factors including:

- The Pre-roll setting (if streaming is enabled)
- The Sample Buffers setting (if streaming is enabled)
- The CPU Cap setting — the amount of CPU resources available to the Emulator X
- The bit depth of your samples — 16 or 24 bits (24 bits = fewer voices)
- The sample rate of the audio hardware — 44.1kHz, 48kHz, 96kHz . Lower sample rates use fewer CPU resources.
- Ultra-High Precision Interpolation and the use of high-order filters can reduce polyphony.
- The speed of your computer — CPU speed, number of CPUs, speed and size of RAM, as well as hard disk access time

Cost of Filters on Voice Count

No Filter	No additional CPU load.
2nd Order	Additional CPU load comparable to playing another 1/2 sample.
4th Order	Additional CPU load comparable to playing another 3/4 sample.
6th Order	Additional CPU load is comparable to playing 1 more sample. (maximum polyphony is divided by 2)
12th Order	Additional CPU load comparable to playing 2 more samples. (maximum polyphony is divided by 3)

Synthesizer parameters also use CPU resources. When creating presets, feel free to use modulation cords as needed, but don't leave cords half connected if you aren't using them, since these patch settings waste CPU cycles. **Turn off both the source and destination of unused cords.**

Audio Setup

These controls let you set audio quality parameters related to Emulator X and your sound card.

Type

You have a choice to use either Direct Sound or ASIO audio drivers. If you have another high quality sound card installed in your system, it will be shown in this field.

ASIO (Audio Stream Input/Output) is a cross-platform, multi-channel audio transfer protocol. ASIO is fast and supports 16-bit and 24-bit data width. Select **E-MU ASIO** if using an E-MU Digital Audio System. Select **ASIO E-MU USB|0202**, **ASIO USB|0404** or **ASIO E-MU Tracker Pre|USB** if using an E-MU USB interface. If you're using another brand of audio interface, select the appropriate driver.

Selecting **Direct Sound** uses the audio output device specified in the control panel of your computer. In general, using Direct Sound is NOT RECOMMENDED because of low transfer rate and high latency issues.

Buffer Size

This control sets the number of sample periods between updates of the synthesizer parameters. The buffer size can be adjusted according to the speed of your computer's CPU. A faster CPU can handle a lower buffer size.

An asterisk (*) next to a buffer size in the drop-down menu indicates the optimal buffer size for your ASIO Buffer Latency settings.

Use Optimal Buffer Size

When this box is checked, the Emulator X will choose the best buffer size based on the ASIO Buffer Latency setting (located in the ASIO control panel). You should normally leave this feature enabled for best audio performance. In certain rare cases you may want to manually set the buffer length to improve MIDI or synth performance.

Sample Rate

This control sets the output sample rate of the Emulator X, and will always match the sample rate of your sound card. The default setting is 44.1kHz.

CPU Cap

This control sets the maximum percentage of CPU resources that can be used by Emulator X and consequently the maximum number of samples that can be played at the same time. (Depending on the design of each preset, playing a single note on Emulator X may cause multiple samples to play simultaneously.) The maximum setting is 80% CPU usage. If you hear notes being "stolen" in your sequence or as you play the keyboard, or if the disk meter readout at the bottom of the window frequently goes into the red, increase the value of this control. If you have other applications running, you may want to reduce the setting of this control to free up more CPU for them.

▼ **Warning:** Sounds with long release times will exhibit "voice stealing" more than short percussive sounds.

Headroom/Boost

Headroom is the amount of dynamic range remaining before clipping occurs.

The amount of headroom is adjustable from 12 dB to -30 dB in 1 dB increments. A headroom setting of 12dB provides the hottest output level, (and the highest signal to noise ratio) but may produce "clipping" if too many notes are played at once. The default headroom setting is -7 dB, which maintains an excellent signal to noise ratio while keeping a reasonable amount of headroom in reserve. If you hear the signal clipping or breaking up, make the headroom value more negative. In practice, you can think of this control as adjusting the control range of the Master Volume control.

Ultra-High Precision Interpolation

This button allows you to select between E-MU's ultra-high precision pitch shifting algorithm or a lower quality pitch shifting scheme which uses fewer CPU resources.

If you are using presets with a sample placed on every key, as is the case on many banks, you can switch high precision off to free up CPU with no penalty in audio quality. No pitch shifting is occurring anyway. High quality pitch interpolation is only needed when you are playing presets which have a few samples mapped across the entire keyboard. In this case, Emulator X shifts the pitch of the available samples to fill in the keyboard.

ASIO Control Panel

This button brings up the E-MU ASIO control panel which allows you to set the **ASIO buffer latency** in milliseconds. This important setting determines how fast notes will sound after you play them. If this time is too long, (>10mS) the keyboard response will feel slow and sluggish. If set too fast, however, you will hear the sound break up and crackle when you play too many notes. Experiment to find the setting that gives the best performance on your computer.

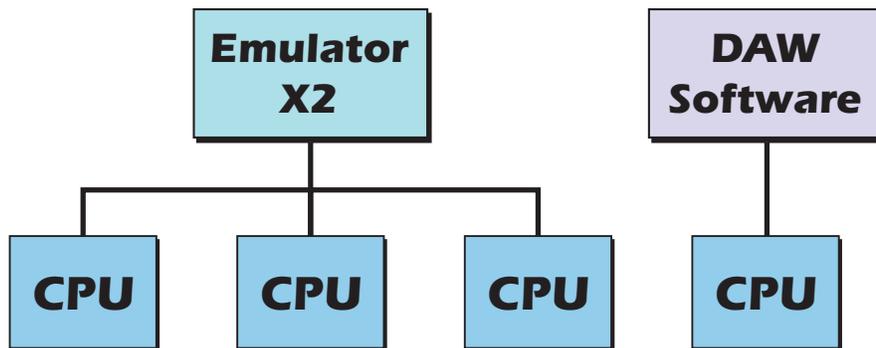
Use Multiple Processors

This option allows the processing load of the Emulator X to be divided among several CPUs or multi core processors. You can also set the **Number of Processors** (up to 4) to utilize the latest quad-core processors for significant performance gains. You can even use the 2 processor option if you have a hyper-threaded CPU since it can run 2 simultaneous threads.

If your computer is also serving as your DAW, recording audio tracks and running VST plug-ins, simply assigning all your available processors to the Emulator X may not always be the best approach. Since your computer is running many other processes besides the Emulator X, you want to balance the workload between your available cores for most efficient results.

For example, if you have a screaming fast quad-core system you might assign the Emulator X to use 3 of the 4 available CPUs for a significant performance gain without crowding out your DAW. On the other hand, if you have a computer dedicated to the Emulator X you might want to give it all the available CPUs.

■ **Tip:** To find out how many logical processors your computer supports check the "Performance" tab in the Windows Task Manager (Ctrl-Alt-Delete) or "Processors" in the Device Manager.



When running multiple audio software applications, it's usually best to balance the workload between the available CPU cores.

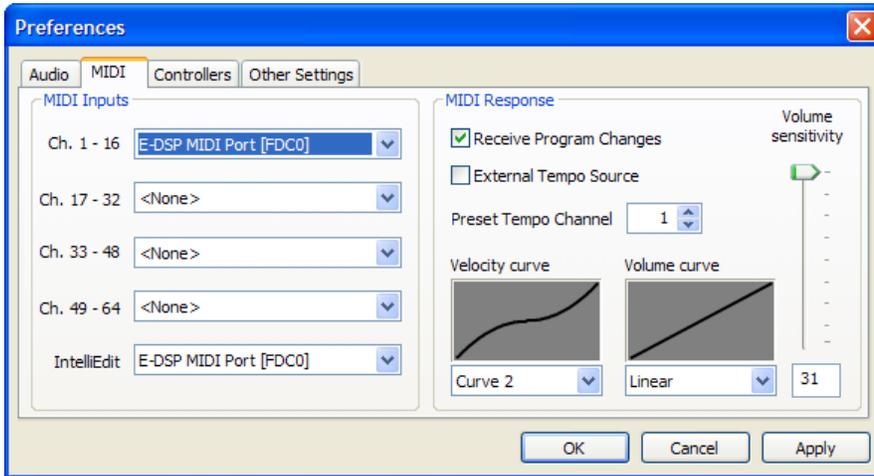
Balancing the workload between the available CPUs is the key to maximizing the performance of your system. If you have a multi core machine and Multiple Processor mode is OFF, you can probably turn up the CPU Cap (try 80%) for a performance gain, since the multiple CPUs are sharing the overall task load.

VST Output Buses

This control is only active in the Emulator X VSTi application. This setting allows you to add additional VST buses (stereo pairs). Emulator X channel outputs can be assigned to VST buses in the Multisetup, Output field. These outputs are sent to the VST Mixer in your recording application. You must restart the VSTi for the changes to take effect.

MIDI

This group of preferences allows you to set up the MIDI inputs, adjust the master MIDI volume and velocity curves and turn the “IntelliEdit” feature on or off.



MIDI Inputs

Channels 1-16, 17-32, 33-48, 49-64

Emulator X can respond to 64 MIDI channels. Since there can be only 16 MIDI channels per MIDI cable (or input), four inputs are provided. Set the MIDI inputs to match up to four outputs from your MIDI interface or sequencer. If you don't need 64 channels you can turn any input off.

IntelliEdit

This control allows you to set the MIDI device for IntelliEdit. Selecting “None” disables the feature.

When editing sample key ranges or editing voices it is convenient to select the key position by simply playing your MIDI keyboard. IntelliEdit lets you selectively edit key ranges using the keyboard when you need to, but blocks keyboard editing when you may not want it, such as when your keyboard is playing in the background.

► To Use IntelliEdit:

1. Make sure IntelliEdit is set to the same MIDI port your keyboard is using.
2. In one of the Voices and Sample Zones screens such as “Key Window”, position the cursor inside one of the Key Range fields (Low or High).



3. Press and hold, **Ctrl+Alt** on your computer keyboard.
4. Play your MIDI keyboard. The key range will be edited.
5. In the Voice Edit window, holding **Ctrl+Alt** lets you select voices for editing. If more than one voice is assigned to the key, the voices will cycle around with repeated pressing of the same key.
6. To select multiple voices, you must use the Group feature. [See Groups on page 93.](#)

Receive Program Changes

This check box selects whether or not MIDI Program Change messages from your MIDI controller or sequencer will be received or ignored by Emulator X. Place an X in the check box to receive program changes.

External Tempo Source

This check box selects between internal tempo clock or MIDI clock as the tempo source. Emulator X uses a global master tempo for Tempo-based Envelopes, Tempo-based LFOs, clock modulation and TwistaLoop. When this check box is selected, MIDI clock will be used for the global tempo and the tempo control in the Multisetup window will be disabled.

Preset Tempo Channel

The tempo of Emulator X can be specified in the preset so that selecting a preset on a specific MIDI channel changes the tempo. [See Initial Tempo on page 82.](#) This option selects the MIDI channel whose preset will be used for Tempo Control. This allows the proper tempo to be set when a Tempo-based or TwistaLoop preset is selected.

For example, suppose you have this field set to channel 1 and the preset on MIDI channel 1 has its Tempo Control set to 122.22 BPM. When the bank is loaded, the Tempo will be set to 122.22 BPM. If you change the preset on channel 1, the tempo will change to follow the "Set Global Tempo" setting in the new preset.

If the preset on the Preset Tempo channel has its Tempo Control set to "Off", (or if this option is set to "Off") no tempo change occurs when the preset is selected.

MIDI Response

These controls allow you to customize the MIDI response of Emulator X to match your MIDI Controller or other instruments.

Velocity Curve

Incoming velocity values can be scaled by one of 24 curves to better adapt to your playing style or MIDI controller. Selecting "linear" leaves incoming velocity data unaltered. The shape of the selected curve is displayed in the window. Select the curve that works best for you.

Volume Curve

This is an adjustment to help match MIDI controller #7 volume response to other manufacturers' equipment. Three curves are provided: Linear, Inverse Square or Logarithmic. The action of this control is displayed in the window.

Volume Sensitivity

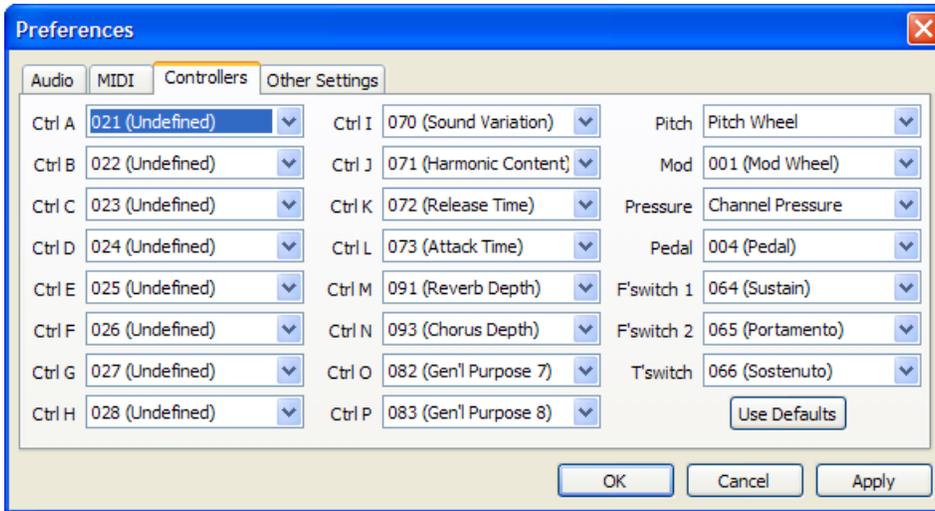
This control allows you to modify the response curve for MIDI continuous controller #7 (Volume). This allows you to match Emulator X's response to other manufacturers' equipment. Low numbers compress the volume control range, raising the volume level at lower controller #7 values.

▼ Important!

If External Tempo Source is set to external, the tempo control in the Voice Processing window will be "greyed-out" and non-functional.

Controllers

This is where you set up which MIDI Continuous Controllers Emulator X will receive. Match these controls to the MIDI continuous controller numbers that your keyboard or sequencer transmits. What the controllers actually do is programmable in each preset. [See Modulation Cords on page 156.](#)



As you can see, MIDI Continuous Controller numbers are assigned to letters (A-P) or labels (Pitch, Mod, Pressure, Pedal, etc.) in this screen. When programming a preset, these letter or name labels can be assigned to control various parameters such as filter frequency or attack time.

The name labels such as pitch and mod wheels are so commonly used that they have their own assigned name, but these can also be freely assigned in any way you choose.

If you're just getting started, it's probably best to use the default settings. Press the "Use Default" button to restore the default settings shown above.

◆ For more information on MIDI Controllers...
[See MIDI Channels & Real-time Controls on page 314.](#)

Controller Knob Assignment

MIDI Keyboard

Controller Knob data

Preferences
(Options, Preferences Controllers)

Voice Processing Cords

1. Select the CC numbers you wish to receive.

Continuous controller numbers are assigned Letters (A-P) & Labels (mod wheel, etc.)

2. Assign the controller letters to Synth destinations.

The Controllers screen determines which CC numbers (knob controller data) the Emulator X will receive. What the knobs actually do to change the sound can be programmed differently for each preset (Voice Processing).

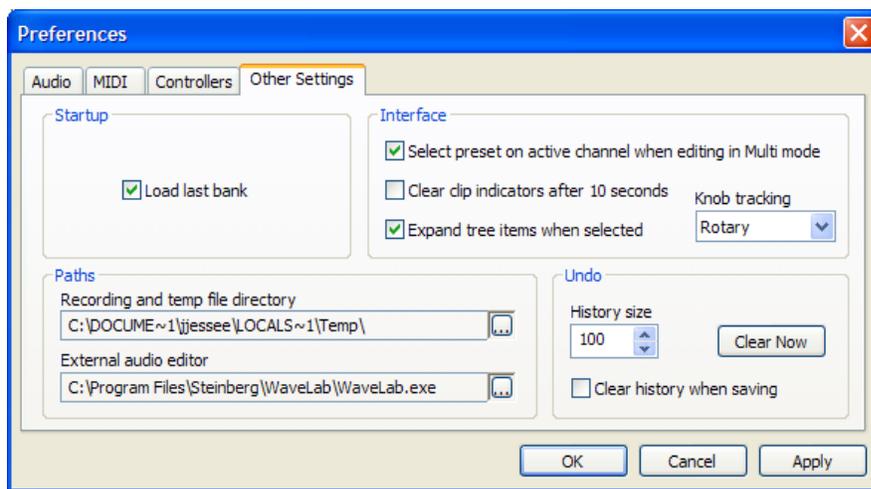
Appearance Settings in Windows

Adjust the “Performance Options” in Windows to make the Preference buttons much easier to read and increase smoothness when dragging windows around the screen.

► To Improve the Appearance Settings:

1. Open the Windows Control Panel. (*Start, Settings, Control Panel*)
2. Select System.
3. Select the **Advanced Settings** tab.
4. Under Visual Effects, select **Adjust for Best Performance**.
5. Click OK.

Other Settings



Startup

- **Load last bank** - This feature automatically loads the last bank you had loaded whenever you start the Emulator X application.

Interface

- **Select preset on active channel when editing in Multimode** - When checked, selecting a preset for editing in the Tree selects that preset on the active channel in the multisetup. This is what you want if you're editing a preset and playing a MIDI keyboard.
If you're playing back a multi-timbral sequence and want to make a quick edit to a preset, you might want this feature Off so as not to alter your multisetup.
- **Clear clip indicators after 10 seconds** - This feature automatically clears the main output meter clip indicators. If this function is set to Off, the clip indicators will stay on until you manually click on them.
- **Expand tree items when selected** - If this option is checked, selecting presets from the Tree will automatically open the selected preset for editing. When unchecked, selecting presets from the Tree will select the preset for playing, but not open it for editing.
- **Knob Tracking** - This control allows the rotary knobs to be manipulated with either Rotary, Horizontal, or Vertical motion. The setting affects all rotary controls.

Paths

- **Recording and temp file directory** - This option allows you to choose the location where new samples will be stored when sampling.
- **External audio editor** - This option allows you to set the location of an external audio editor. The External Audio Editor is selected from the Sample Edit menu or the toolbar.  When the External Editor is selected, the current sample opens in the editing application you have selected. When you Save the edited sample in the external editing application, it is automatically transferred back to the Emulator X. You even have the option to Undo the external edit.

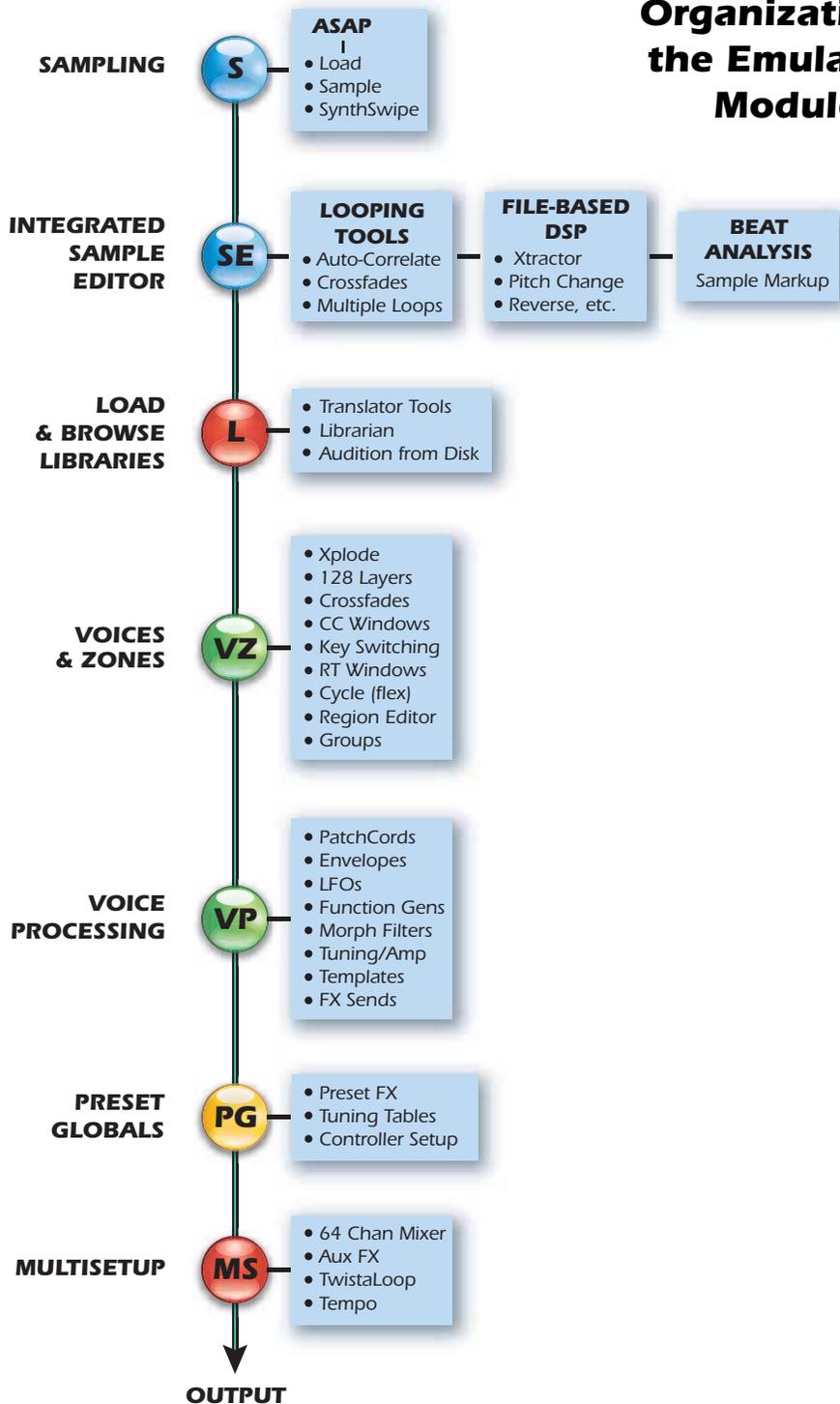
Undo

- **History size** - The maximum number of stored undo steps.
- **Clear history when saving** - Clears all undo history when you save a bank. If you have this box unchecked, you can undo even when you save and re-open a bank.
- **Clear now** - Clears the undo history immediately.

2 - Emulator Architecture

This chapter contains important background information about the organization of the various modules of Emulator X. Although this chapter doesn't contain any hands-on tutorials, this information is vital to your understanding of this ultra powerful instrument. Read This!

Organization of the Emulator X Modules



A Modular System

You can think of Emulator X as a collection of sound organizing modules, all contained within the current bank. The following is a brief description of the five main elements of Emulator X's hierarchy starting from the largest element, the Bank.

Bank

Before you can play Emulator X, you load a *Bank* of presets. The bank contains all the sounds you wish to use in a particular sequence or performance. If you need additional sounds, they can be added by simply dragging and dropping them into the bank from the system or library. Unneeded sounds can likewise be removed from the bank. When loading a bank, Emulator X only loads a small portion of the digital samples so bank load time is greatly reduced.

Saving

A bank's data is retained only for as long you have the program open. Saving the bank to disk permanently stores the bank so you can access it next time you use the program.

IF YOU DO NOT SAVE A BANK, ANY CHANGES YOU HAVE MADE WILL BE LOST WHEN YOU OPEN A NEW BANK, OR CLOSE EMULATOR X!

Do not wait until the end of a session to save. Save your work periodically in case of power failure or some other unforeseen circumstance that might exit the program. Hard disks and computers are not infallible. All hard disk banks should be backed up periodically to another hard disk or other media. Should you improve the preset or sample later, you can always replace the original with the revised version. And if something goes wrong, the original will still be available to save you the ordeal of starting from scratch.

When you choose **Save**, (as opposed to **Save As**) only the edited presets, voices and samples will be written, saving time and memory.

When you choose **Save As** from the **File** menu, the entire bank—presets, voices and samples—are re-written to the hard disk as new files. This method, although somewhat wasteful of memory, ensures that all your samples remain bound with your bank.

When you save, you might be prompted with a dialog box telling you have unused samples. Choosing to delete the unused samples removes them from your hard drive, saving space and memory. This dialog occurs any time you have samples in your sample pool that are not assigned to any voices.

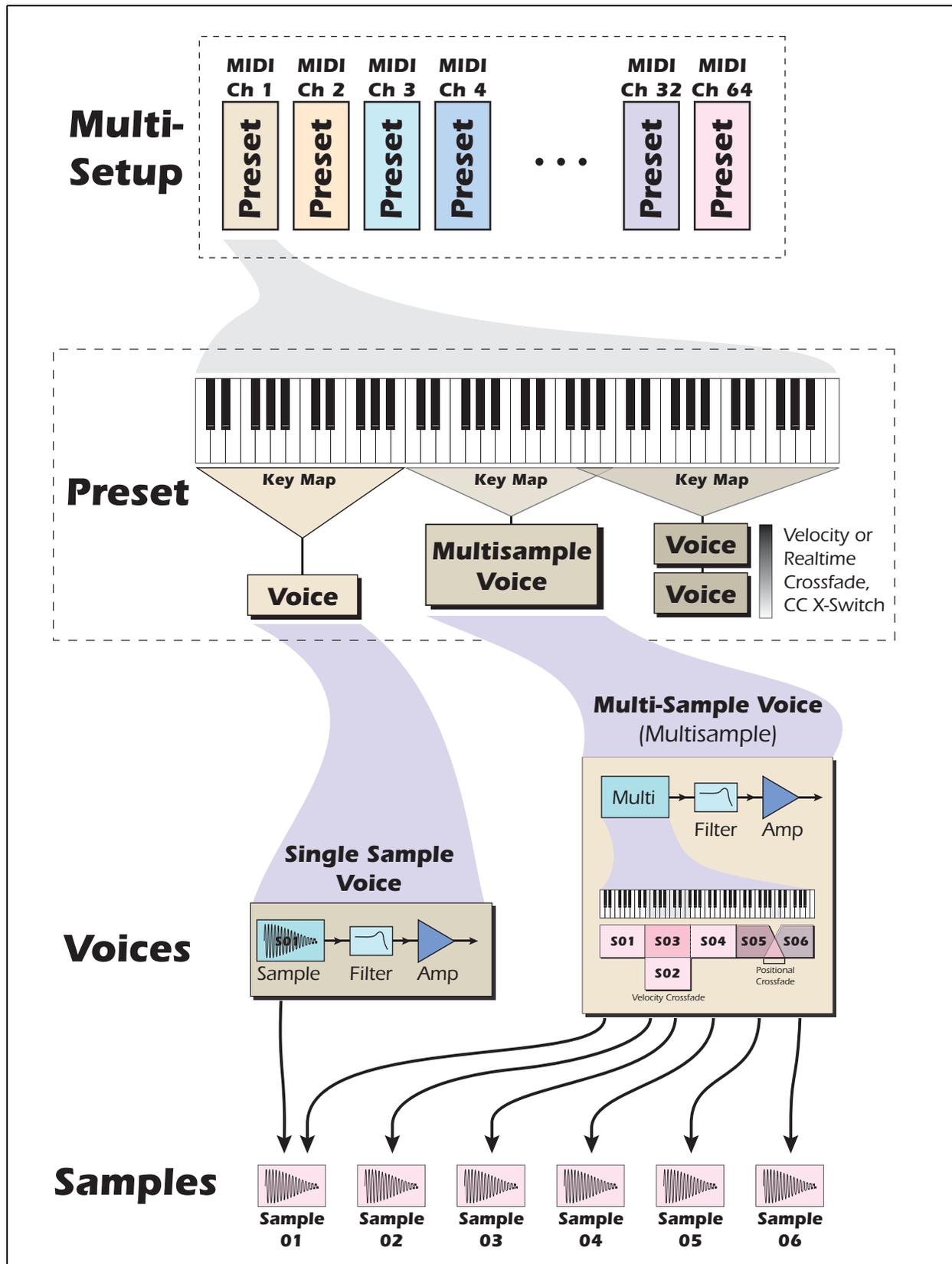
Undo/Redo

Almost every action you do with Emulator X can be undone. You can undo up to 500 actions and this number is selectable under **Preferences** in the **Options** menu. See [page 30](#). Select **Undo** or **Redo** from the **Edit** menu or press **Ctrl+Z** (**Undo**) or **Ctrl+Y** (**Redo**) from the keyboard.

▼ Whenever you have done enough work that you would hate to lose it, **BACK IT UP!**

Use **Save As...** and rename the bank file to avoid losing valuable data.

Emulator X Bank



Multisetup

A Multisetup is like a map which assigns a preset, volume, pan position and output routing to each of the 64 MIDI channels. Aux FX and effect bypass controls are also included in a Multisetup. Multisetups can be saved and recalled for use with a particular song or sequence so that all the MIDI channels play the proper preset.



Preset

A Preset is one complete keyboard setup controlled by one MIDI channel. Presets are composed of multiple voices. The assignment of voices to keyboard keys is completely flexible.

◆ A Preset is the same as a MIDI Program.



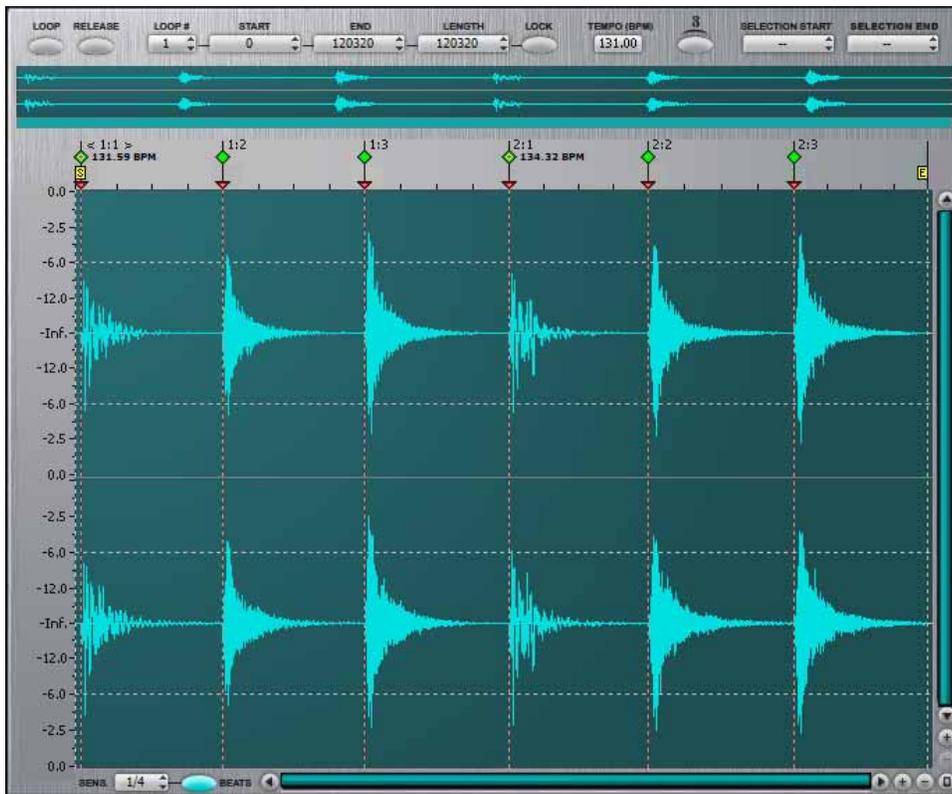
Voice

A Voice is one complete sound, containing one or more samples with keyboard and velocity mappings and all programmable synthesizer parameters. Voices can be assigned to a single note on the keyboard, or transposed to cover a wider keyboard range.



Sample

A Sample is an individual digital recording with a name, sample rate and looping information.



Flexible Architecture

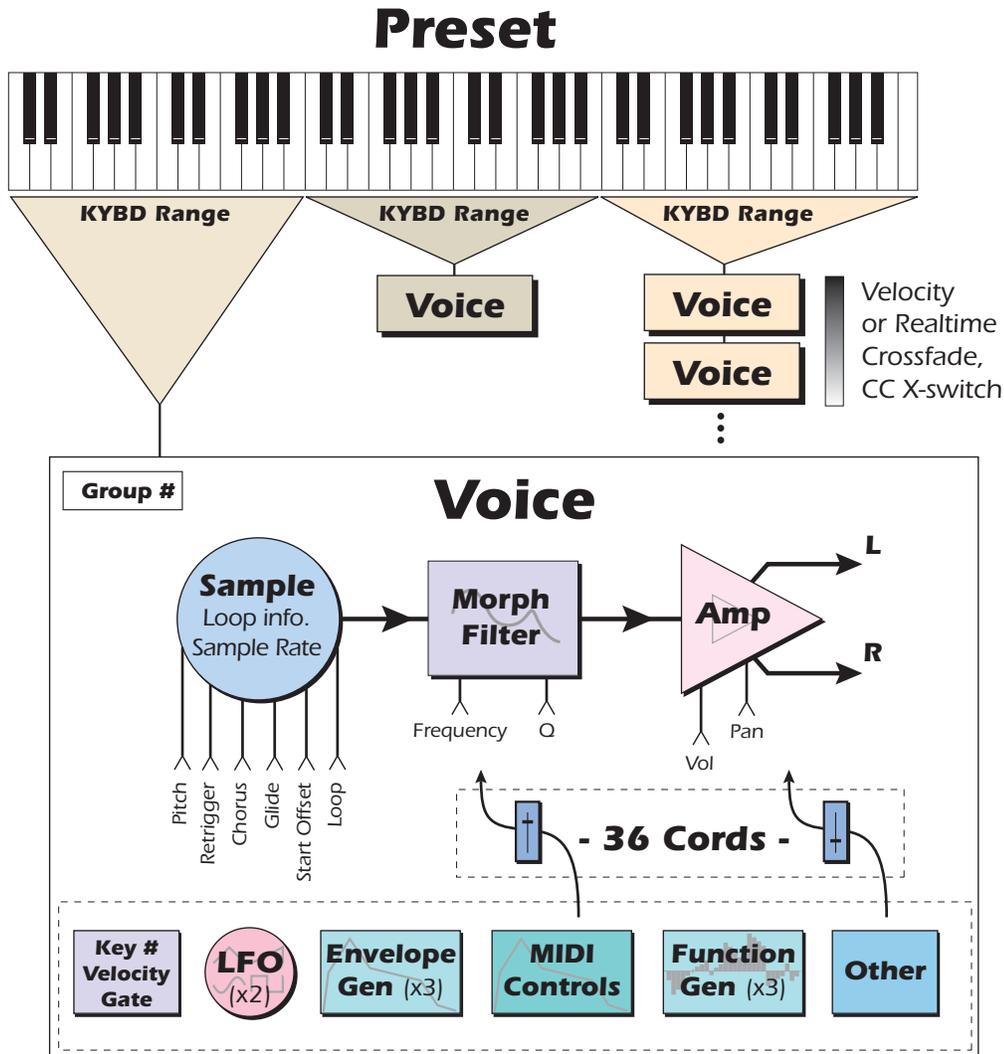
Emulator X allows you great flexibility in the way you construct presets. Consider this — you can assign multiple samples to the keyboard inside a voice or assign single sample voices to the keyboard.

Unless you specify otherwise, only one sample is assigned per voice. In this case you would assign voices (and the single sample each one contains) to the keyboard and create presets. On the other hand, you may want to create multisample voices before you start designing presets and treat the voice as your finished sound. In this case, the Preset can crossfade, layer or switch between multiple complex voices.

Preset Diagram

The diagram below illustrates the concepts of a preset and voice.

- Preset** A complete keyboard layout with unlimited voices.
- Voice** A complete synth sound containing a single sample or a Multisample.
- Sample** An individual digital recording with loop information.



Voices

A voice is a complete sound that can be assigned to a range of the keyboard. A sample is the part of a voice that generates the actual sound. You can think of a voice as a complete instrument consisting of one or more samples, which can then be used as a building block in constructing more complicated presets.

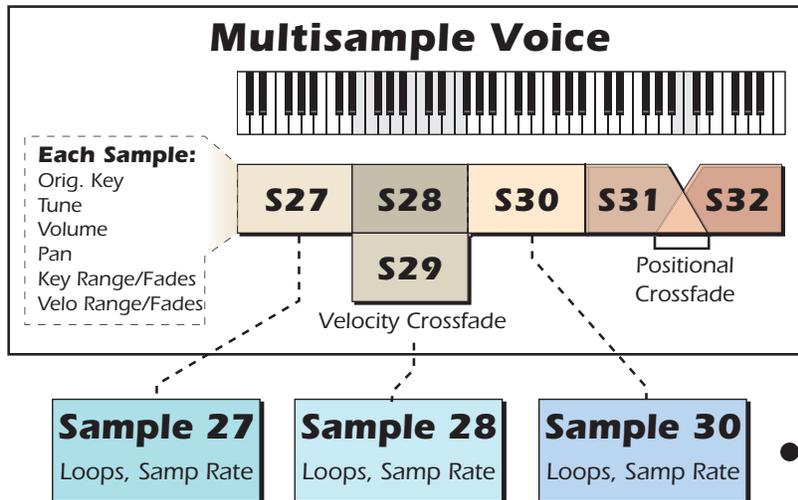
A voice consists of one or more samples, a dynamic filter, a dynamic amplifier, three 6-stage envelope generators, three 64-stage function generators, two multi-wave LFOs and up to 36 modulation routings called "Cords" to connect everything together.

When you record a sample into the Emulator X, a new voice is typically created for each new sample. In this case, each sample will have its own complete set of synthesizer parameters.

Multisample Voices

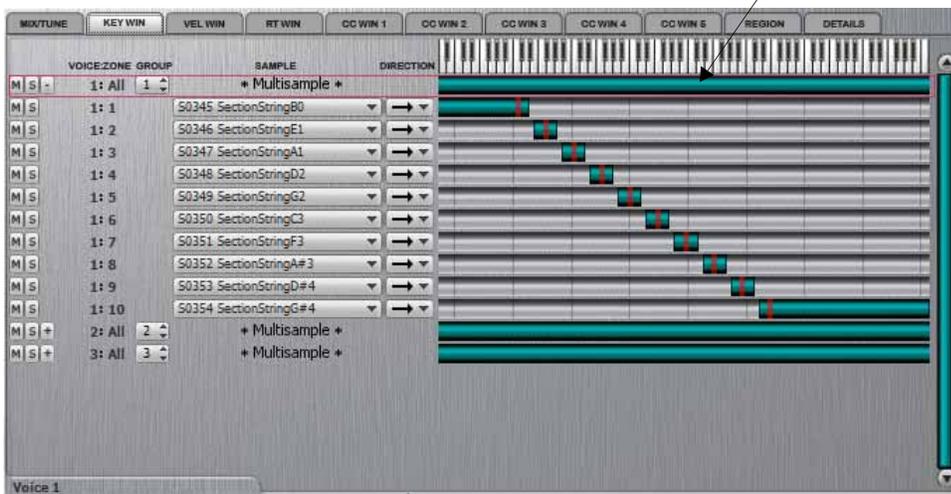
In another scenario, you might record several samples of an instrument (such as a piano), then place them into the same voice in order to share the same set of synthesizer parameters. If a voice contains more than one sample, this multiple sample object is called a "Multisample." A diagram of a multisample voice is shown below.

- A voice can contain a single sample or multiple samples, whichever you prefer.



Multisample voices are designed to arrange groups of samples into one manageable entity.

The window below shows an opened multisample. The key range of the multisample can be used to limit the keyboard range of the samples contained within it. A multisampled voice is treated just like a single sample voice.



Normally these samples would be placed side by side on the keyboard as in the diagram above. You assign the sample to a range by setting the original key, (which is usually the original pitch of the sample) a high key and a low key. The number of samples needed for a realistic emulation varies with the instrument, but in general, "more is better."

When a sample is taken, it can be automatically placed into a voice with one sample. You could also manually place the voice (and its sample) on the keyboard if you wish. If more than one voice is assigned to the same range, then pressing a key in that range plays all the voices assigned to that range. Voices assigned to the keyboard can be cross-faded by their position on the keyboard, or the key stroke velocity. Voices can also be switched or faded depending on the value of a realtime controller such as a modulation wheel, an LFO or an envelope generator.

The Sample

Loading in any sound, in mono or stereo, creates a sample, the raw material with which the Emulator X works.

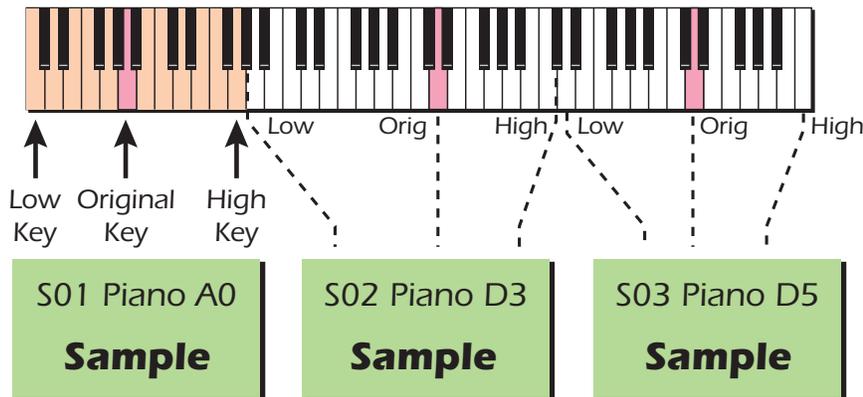
The term sample commonly means two different things:

- A digital recording of a complete sound, or
- Each snapshot of the sound that makes up the complete sample.

In this manual, we'll assume sample means the complete recorded sound unless indicated otherwise.

You can modify a raw sample in several ways. A sample can be transposed up or down in pitch to cover a particular range of the keyboard making it unnecessary to record a sample for every key. Sample editing might consist of *Looping* a sample (allowing even short samples to play indefinitely), *Truncating* (cutting off unneeded parts of a sample), or any of a number of digital processes that actually change the raw sample data.

The **Original Key** assignment determines the pitch at which the sample will be played back. Therefore it's important to map your samples to the proper original key.



You can also name samples. It's usually a good idea to include the original pitch in the name so that you can place it on the keyboard later at the proper pitch.

You can have any number of samples in a bank. The number of possible samples is only limited by the size of your computer's hard disk.

TwistaLoop™

TwistaLoop is a tool that automatically analyzes a sample by dissecting it into its rhythmic components, making it easy to select multiple loop points and regions. Twist-aLoop offers total control over tempo, using real-time audio time expansion/compression.

You can play back analyzed samples at a custom speed, or in time with Emulator X's clock (or an external clock source).

Once you create loops from an analyzed sample, you can use them in a variety of ways:

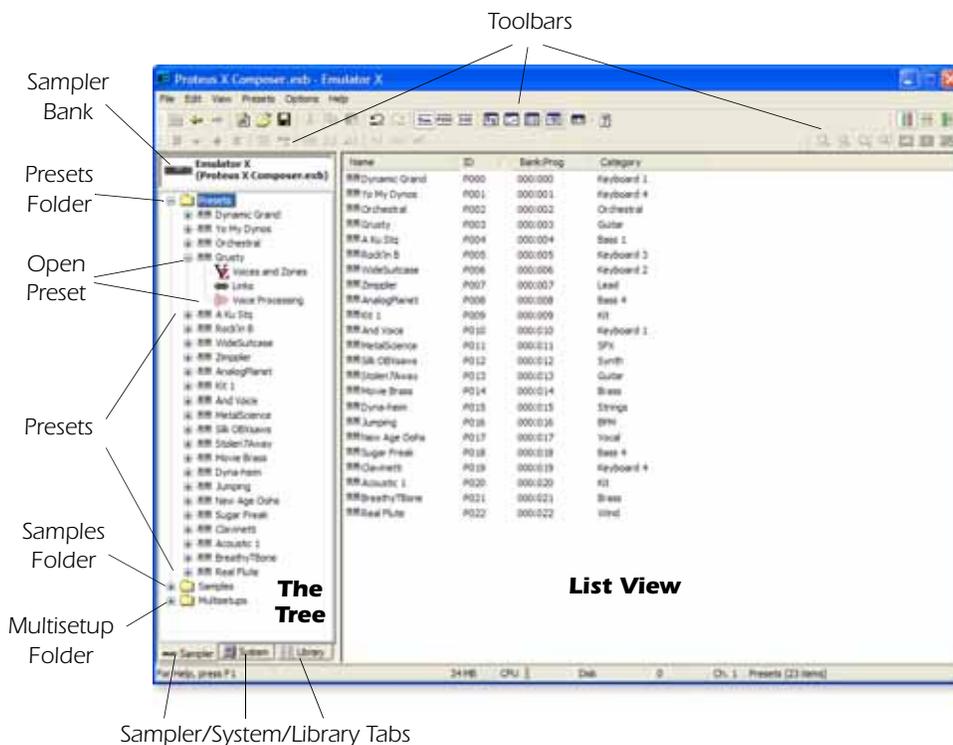
- Change the playback speed of any loop.
- Choose which loops are repeated when played back.
- Set up a continuous controller to change in real time which loop plays.
- Sync your loops to Emulator X's master tempo, or to an external tempo source, such as a VST host.
- Play different beats or bars from the same sample.

3 - Getting Started

Exploring the Tree

In this section you'll learn how to navigate around Emulator X using the Tree. Don't worry if you don't know what everything does just yet. Everything will be explained later on.

1. With the Emulator X application running, a bank is loaded by selecting Open from the File menu. Click on the **Presets Folder** in the Tree and a screen, similar to the one shown below, appears.



The Tree is the section shown on the left of the screen. There are three tabs at the bottom of the Tree labeled **Sampler**, **System** and **Library**.

Sampler Allows you to see the contents of the current Emulator X bank.

System This tab gives you access to your entire computer system: PC, hard disks, CD-ROM drives, network, etc. Presets, Samples and Multisetups can be browsed, then Merged into the current Bank. **Load Emulator 4 banks from this tab.**

Library The Library shows you all Emulator X and related files, wherever they may be located on hard disks, CD-ROM or network. Presets, Samples and Multisetups can be browsed in the Library, then Merged into the current Bank.

▼ E4/E3 CDs use a proprietary format and can only be read using SCSI/ATAPI drives. They cannot be read using firewire and USB drives.

Multisetup Page

The Multisetup is the top level of the Emulator hierarchy and is the place where you choose and assign presets to each of the MIDI channels.

A multisetup assigns a preset, volume, pan position and output routing to each of the 64 MIDI channels. Multisetups can be saved and recalled for use with a particular song or sequence so that all the MIDI channels play the proper preset.

Click on the Emulator X icon to display the Multisetup page.

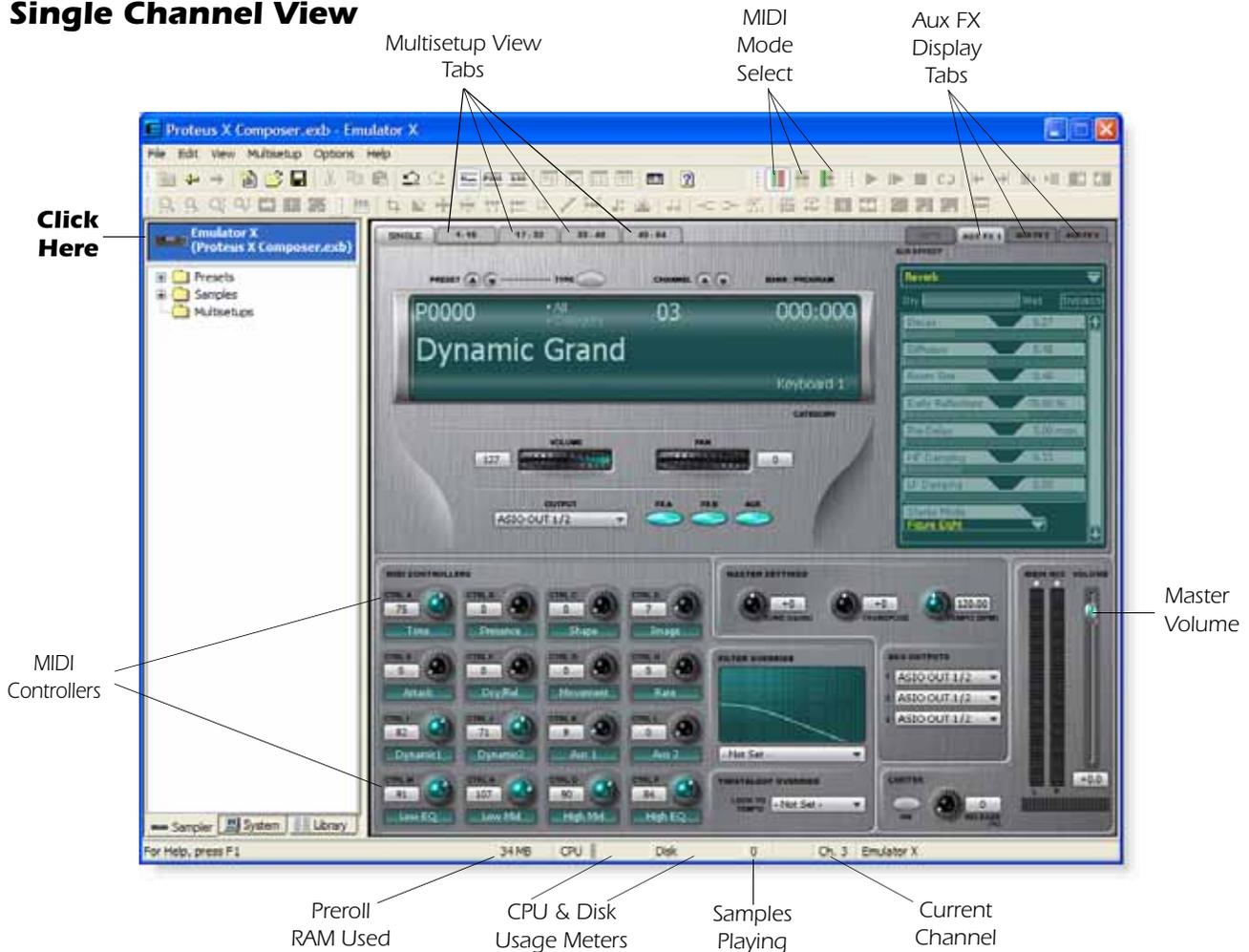
There are five views of the Multisetup page. The **Multisetup View** buttons change the current view. The **Single** channel view is shown below.

- **Single**..... Displays the preset for the currently selected MIDI channel.
- **Channels 1-16**..... Displays the preset, volume and pan for MIDI channels 1-16.
- **Channels 17-32**.... Displays the preset, volume and pan for MIDI channels 17-32.
- **Channels 33-48**.... Displays the preset, volume and pan for MIDI channels 33-48.
- **Channels 49-64**.... Displays the preset, volume and pan for MIDI channels 49-64.

See [page 69](#) for a complete description of the Multisetup screen.

◆ **Note:** When sequencing, you can also include program changes in the sequence rather than using a Multisetup. This is often best, since the program changes will then be part of the song.

Single Channel View



Changing Presets (from Single View)

Single View is the default view when you first run Emulator X. This view is convenient for performing live or when you're only playing one preset at a time. All the main parameters for the selected MIDI channel are displayed in Single View and a convenient Category mode allows you to quickly select presets in a certain category. A close-up of the current channel settings is shown below.

◆ Note that Single View only changes the display and does not affect the currently selected MIDI mode (i.e. Omni, Poly or Multi).

Single View Close-up



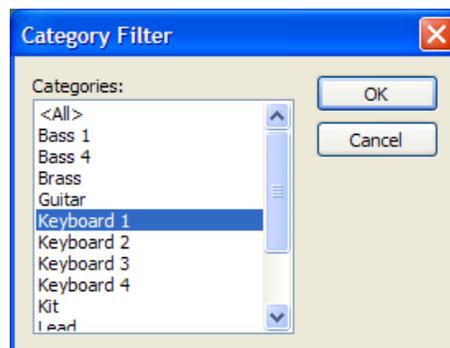
◆ The Bank/Program Display shows the Bank and Program number you would use to select the preset from a sequencer.

► To Select Presets by Number

1. Click on the Preset Increment/Decrement buttons to move up or down the presets by number.
2. Highlight the Preset Number field and type in the desired preset number.

► To Select Presets by Category

1. Click on the Select Category button. The pop-up dialog box at right appears.
2. Select the desired preset category, then press OK.
3. Choose **Select By Category** from the Single view screen above.
4. Use the **Increment** or **Decrement** buttons to select presets. Note that only presets of the selected category are selected.



► **Changing the Current MIDI Channel**

The Current MIDI Channel is simply the MIDI channel you are currently working with. To change the current MIDI channel, simply increment or decrement the channel number using the buttons, or click on the channel number and type in the desired number.

► **To Find the Selected Preset in the Tree**

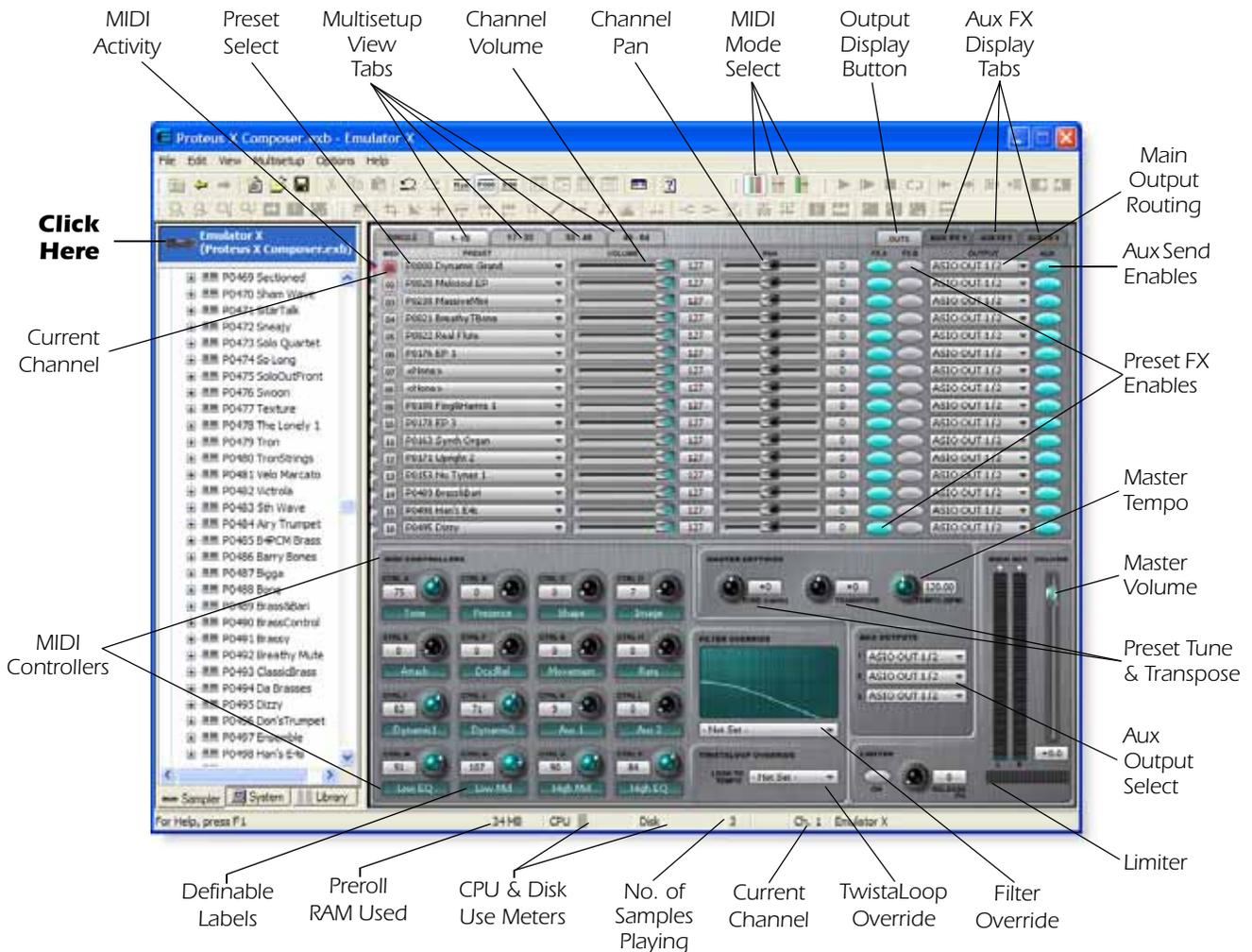
In a huge bank like the Proteus X Composer, you can jump immediately to the preset by **double-clicking in the Preset Name Display**. The Tree expands to that preset, and the view switches to the Preset Globals view.

Multi Channel View

Press the Multisetup View 1-16 tab to view channels 1-16.

This page view is particularly useful when sequencing as you can see 16 MIDI channels at once. Note that the **Output Display** button is now active. Tabs 17-32, 33-48, 49-64 are identical to this page and allow you to access these additional MIDI channels. See also [“4 - Multisetup / Preset Globals” on page 69](#).

Multi Channel View



Changing Presets (from the Multichannel view)

Play your MIDI keyboard and note that one of the MIDI activity LEDs comes on. This is the channel on which your MIDI keyboard is transmitting. If you don't see MIDI activity, check your MIDI connection and make sure MIDI is set up properly in the Preferences menu. See [page 28](#).

► To Audition and Select Presets

1. Click once on the **Preset Select** button on the channel used by your MIDI keyboard. The complete list of all presets in the bank appears.
2. Click once on a **preset** to audition it. Play your keyboard to hear it.
3. Try out a few more presets. When you find one you like, double-click on it to select it and return to the Multisetup screen.
4. Change the current channel by clicking on the number to the left of the preset. The number you selected turns red.
5. Play the floating keyboard (Select **Keyboard** from the **View** menu). Notice that the MIDI Activity LED of the current preset now plays along with whatever sound is selected for that channel. (Your MIDI keyboard still plays on the channel to which it is set.)
6. Change the channel on your MIDI keyboard then play it. Emulator X now plays the preset on that channel.

► Selecting Presets from the Tree

1. When you click on a preset in the tree, you have just selected it for the current MIDI channel. This action also calls the Preset Global page.
2. To go back to the Multisetup page click the **back arrow** or click on the **Emulator X icon** at the top of the tree.

■ Selecting "None" in the Preset field, disables the MIDI Channel.

► Changing the Current MIDI Channel

1. The Current MIDI Channel is simply the MIDI channel you are currently working with.
2. To change the current MIDI channel, click on the channel number in the Multisetup page. The selected channel number turns red. Note also that the channel number at the bottom of the window changes to reflect the current channel.

Adjusting the Controllers

Each preset has a set of 16 continuous controllers (CC's) which are used to adjust and control the sound as you play. Your MIDI keyboard may have several knobs or sliders that are transmitted over MIDI. These can be used to control Emulator X if you match the CC numbers your MIDI keyboard transmits to the same CC numbers on Emulator X. These global settings are located under Options, Preferences, Controllers. See [page 28](#) and [page 314](#) for additional information about MIDI controllers.

► To Modify the Sound using the Controllers

1. Play your MIDI keyboard as you adjust the controller knobs on the screen. Notice that the sound changes. If the sound doesn't change, make sure the current preset (red number) is the same one that your keyboard is playing.
2. If you have controller knobs on your MIDI keyboard and have set up the Controllers (Options, Preferences, Controllers) to match your keyboard, you can use these knobs to modify the sound. Notice that the screen knob follows the movement of the hardware knob.

3. Try changing the filter type. Click on the selector to the right of the filter name and select one of the 55 different filter types.
4. Adjust the Tone and Presence controls while playing the keyboard to hear your changes.
5. Change the Volume and Pan controls for the MIDI channel you are using. These settings can be adjusted remotely using MIDI controller #7 (volume) and controller #10 (pan).
6. If you want to save the changes, you must set the **Initial Controller Amounts** in the **Preset Globals** page to match the settings you made in the **Multisetup** page, then save the bank.

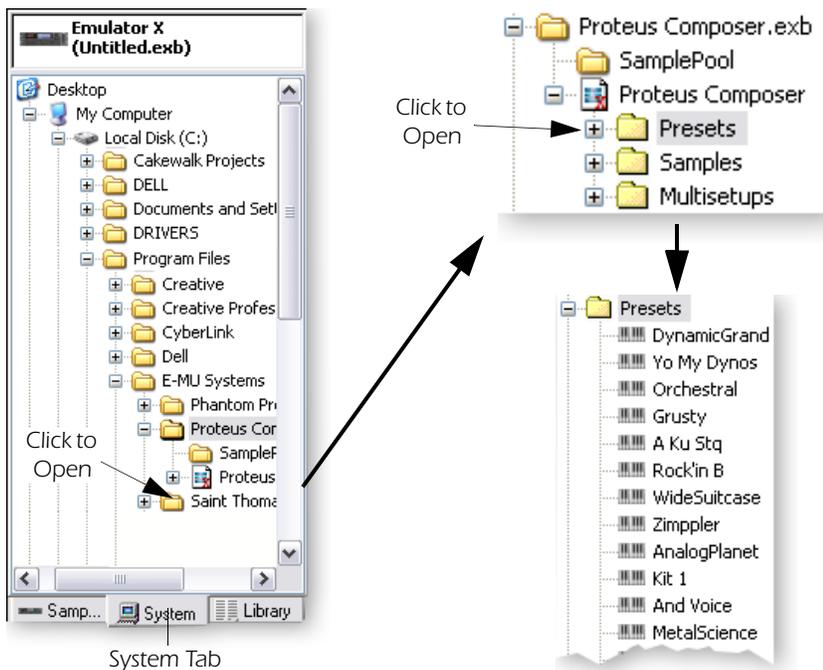
Create Your Own Custom Bank

It's easy to create your own custom sound banks by simply "dragging and dropping" the presets or samples you want onto the Emulator X icon in the tree.

► To Create a New Custom Bank

Locate the Presets on your Computer

1. Select **New** from the File menu. This creates a new empty bank. (You can also merge presets or samples into an existing bank if you wish.)
2. Select the **System Tab**. The Desktop icon will appear in the tree.
3. Locate the presets you wish to include in your new bank. These will be located in the "Creative Professional" folder located in "Program Files".



■ The included CD-ROMs contain several other great banks of sounds. Run the installer application on the library CDs to install these banks on your hard disk.

Audition the Presets

4. After you have located the **Presets** folder, click on the **Emulator X** icon in the Tree to view the Multisetup page.
5. Find a preset in the Tree and **Right-click** over it. A pop-up menu appears with two options, **Audition** and **Merge**.
6. Select **Audition** (or press "A" on your computer keyboard). Notice that the preset name is now flashing in the Multisetup screen.
7. **Play your MIDI keyboard** to audition the preset from the hard disk.
8. **Right-click** again and select **Stop Audition** (or press "S" on your computer keyboard). The preset name stops flashing in the Multisetup screen.
9. Continue auditioning until you find a preset you like.

■ **Tip:** You can also audition samples or presets from the library.

◆ Presets that reside in .E4B or .EXB disk files can be auditioned from disk.

Merge the Preset

10. When you have found a preset you want to include, **Right-click** over it in the Tree and select **Merge** from the menu (or press "M" on your computer keyboard). The preset is added to the current bank.

Another Method of Merging Presets

11. Simply **click** (left mouse button) and **drag the preset** over the **Emulator X icon** at the top of the tree. The cursor changes to a plus sign as you pass over the icon. Release the mouse button and a pop-up progress bar will inform you that the preset is being loaded.
12. If you change your mind and don't want to merge the preset after all, choose **Undo** from the Edit menu to un-merge the preset.

Continue to Create the Bank

13. Continue to add presets to your new bank.
14. Save the bank when you are satisfied.

► **Alternate Method using the Librarian to find Presets**

1. Select the **Library** Tab instead of the System Tab in step 3 above.
2. If you haven't done so already, **Update** the Library.
3. Choose presets from the Library, then **drag and drop** them on top of the **Emulator X icon** at the top of the tree.

Load a New Bank

Load up a new bank in preparation for the next few examples. Load the Proteus X Composer bank located in your Emulator X folder.

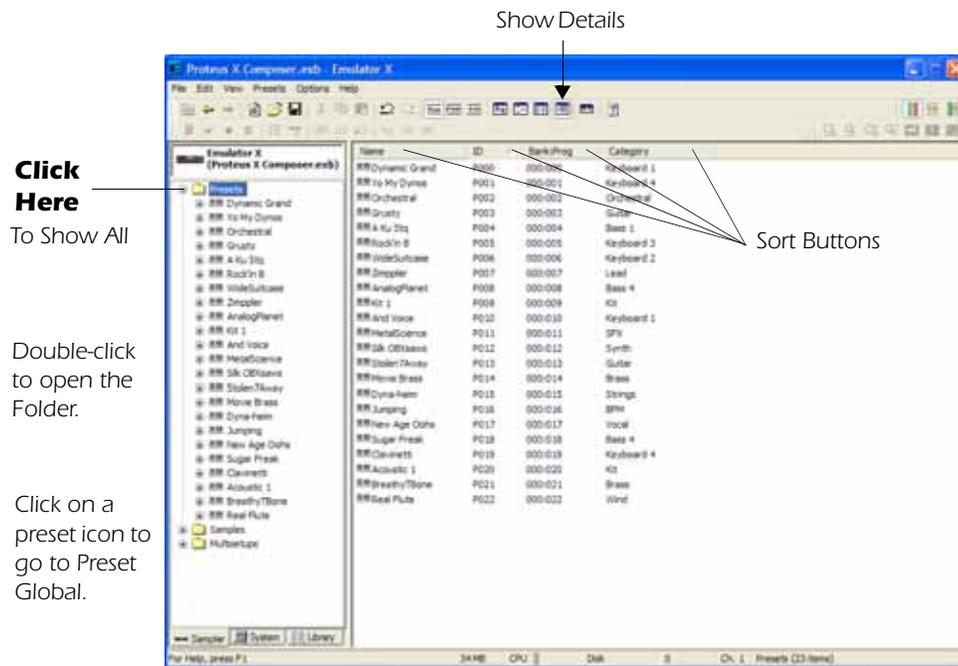
1. Select Open from the File menu, then locate Proteus X Composer and click Open.

◆ **Note:** Emulator 4 banks are accessed and loaded from the **System Tab**, rather than from the File menu.

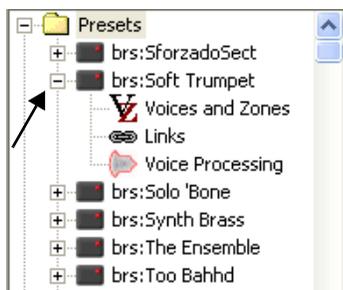
Opening a Preset

A preset is a complete keyboard setup for one MIDI Channel. Think of a preset as one complete sound, which usually contains multiple voices and samples.

1. Click on the Preset Folder in the Tree to show all the presets in the bank.



2. Click the Show Details button. The details of the presets are now shown. The sort buttons allow you to sort the presets by name, ID number, bank/program number, or category. Click on the button again to sort in reverse order.
3. Clicking on the plus sign (+) next to the Preset folder or double-clicking on the folder itself opens the folder and displays the bank's presets in the tree.
4. Clicking on the plus sign (+) next to one of the preset icons displays the sub-modules of the preset: Voices & Presets, Links, Voice Processing.



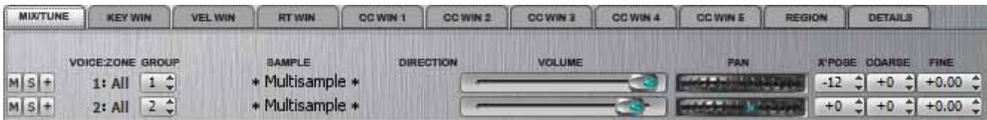
◆ Please refer to chapter [“2 - Emulator Architecture” on page 3.1](#) for complete explanations of the sub-modules.

Submodule	Function
Voices & Zones	Assign voices and multisamples to the keyboard, which can be switched or crossfaded by key position, velocity or controller setting.
Links	Multiple presets can be layered, switched or crossfaded across the keyboard,
Voice Processing	Synthesizer voice parameters including filters, LFOs, envelopes, function generators and cords.

- Open the preset “And Voice” by clicking on the plus (+) sign next to the folder.
- Next, click on the **Voices & Zones** icon. The Key Window appears. The blue bars show that the two multisamples in this preset are mapped across the entire keyboard.



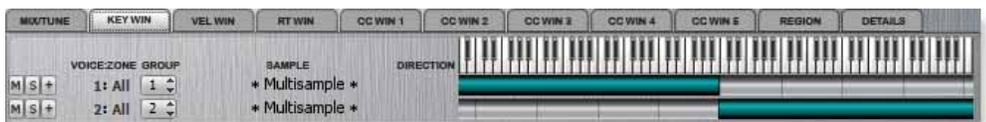
- Click on the **Mix/Tune** tab at the top of the window.



- There are two multisampled voices in this preset. Adjust the volume slider for each of the voices as you play. Now try playing with the pan position and transpose. Before you leave this window, turn up the volume controls on both voices.

► Exploring the Key and Velocity Windows

- Click the **Key Win** tab again.



- Click and drag the ends of the blue bars so they look something like the screen shown above. The settings don't have to be exact.
- Play the keyboard from the lowest key to the highest. Voice 1 plays at the low end of the keyboard and voice 2 plays on the high keys.
- To return both voices to their original positions, select **Edit, Undo** from the toolbar. Select **Undo** again until both voices are assigned to the entire keyboard.

Velocity Switching and Crossfading

- Click on the **Vel Win** tab. The following window appears:



6. This page controls the voices using key velocity or how hard the keys are played. Set the bars something like the screen above.
7. Play the keyboard anywhere, but begin playing softly and gradually play harder. Notice that the choir (voice 2) plays with lower key velocity and switches to piano (voice 1) when you play hard. You've just created a velocity cross-switch.
8. **Alt-click** the end point of the bar in voice 1 and drag it to the left. The color intensity of the bar fades. The fade in color is analogous to volume. the darker the color, the louder the volume for a certain velocity.
9. Drag the fade points of the two voices so they look something like the window shown below.



10. Play the keyboard anywhere, but begin playing softly and gradually play harder. Notice that the choir plays with lower key velocity and fades into piano as you play harder. You've just created a velocity crossfade.

► **Add a Voice**

1. Select **Preset, New Voice** from the toolbar. A new, empty voice appears below voice two.
2. The label area of the new voice is blank. Click on the **selection box** to the right of the empty label area to bring up the entire list of samples in the bank.
3. Select sample **0350 SectionStringC3** by double-clicking on it. You've just assigned a sample to the voice you created.
4. Play the keyboard and now you have orchestral strings as well.

► **Delete the Voice**

1. Select the voice you just created by clicking once on the **sample name** or **number** to the left of the name. The number turns red.
2. Select **Preset, Delete Voice** from the toolbar. The voice has now been deleted.

Examine a Voice

Each voice has its own synthesizer section (called Voice Processing) to process the raw samples. You can process each voice separately or select all voices to be processed in the same way.

1. Go to the Tree and click on preset P0015 Dana-him. Play the keyboard. You should be able to hear the preset.
2. Open the preset by clicking on the **plus sign(+)**, then click on the **Voice Processing** icon. 🖱️ The Voice Processing page shown below appears.



Step 3.
Select All

Step 5
Set to LFO 1 ~

Step 6
Set to Filter Frequency

3. Near the top of the screen near the keyboard you'll see the **Group Selector**. Set this to **All**, to select all voices in the preset.
4. Feel free to explore the synthesizer section. The filter section is probably a good place to begin. Change the filter type, then adjust the frequency and Q controls as you play the keyboard. Note that the filter response display changes as you change the filter settings.

◆ See Groups on [page 93](#) for detailed information on how to select groups of voices.

Cords

One reason synthesizers are able to generate such complex sounds is because most, if not all, of their processes can be placed under automatic control. Here's an example.

5. Tired of turning the filter frequency by hand? Let's program one of the LFOs to do it. Choose Cord #8 and set the left selection box to **LFO 1 ~**.
6. You've just connected one end of a Cord. Like all cords, you've got to connect both ends for it to do anything. Connect the other end of the cord to **Filter Frequency**.

7. Each Cord has its own attenuator which controls “how much” signal goes through the cord. Turn the knob in the center of the cord all the way to the right (+100%).
8. Now set the filter frequency knob about 1/3 of the way up and play a note on the keyboard. You should hear the filter being turned up and down automatically.
9. Change filter types and adjust the filter frequency if you don’t hear the filter being modulated.
10. Adjust the **Frequency** of LFO 1.
11. Change the **Shape** of LFO 1. (You’ll have to re-key to hear the new waveform.)

■ **Tip:** Any control can be set to its default value by pressing **Alt + left mouse click** over the control.

Modulating Cords

In Emulator X even the Cords themselves can be controlled by other modulation sources or controls. Let’s use your keyboard’s mod wheel to control the amount of LFO to filter modulation.

12. Cord 0 already has the Mod Wheel assigned as an input. Set the output of **Cord 0** to go to **Cord 8 Amount**.
13. Turn up the amount of **Cord 0** to **100%**.
14. Set the amount of **Cord 8** to **0%** (off). (*You’re going to turn it up with the mod wheel.*)
15. Now play a note and move the Mod Wheel of your keyboard up. (*Most keyboards have two wheels or controllers. One bends pitch and the other is the modulation wheel.*)

Experiment

16. Try setting the output of Cord 11 to **Pitch**. Now try changing the LFO Shape. Now it’s much easier to hear the different LFO wave shapes.

Wipe the Slate

The preset might be a little messed up after your experiments. It’s good to know that as long as you don’t **Save** the bank, none of the original data is destroyed. If you want to get back to the original preset, simply load the bank again.

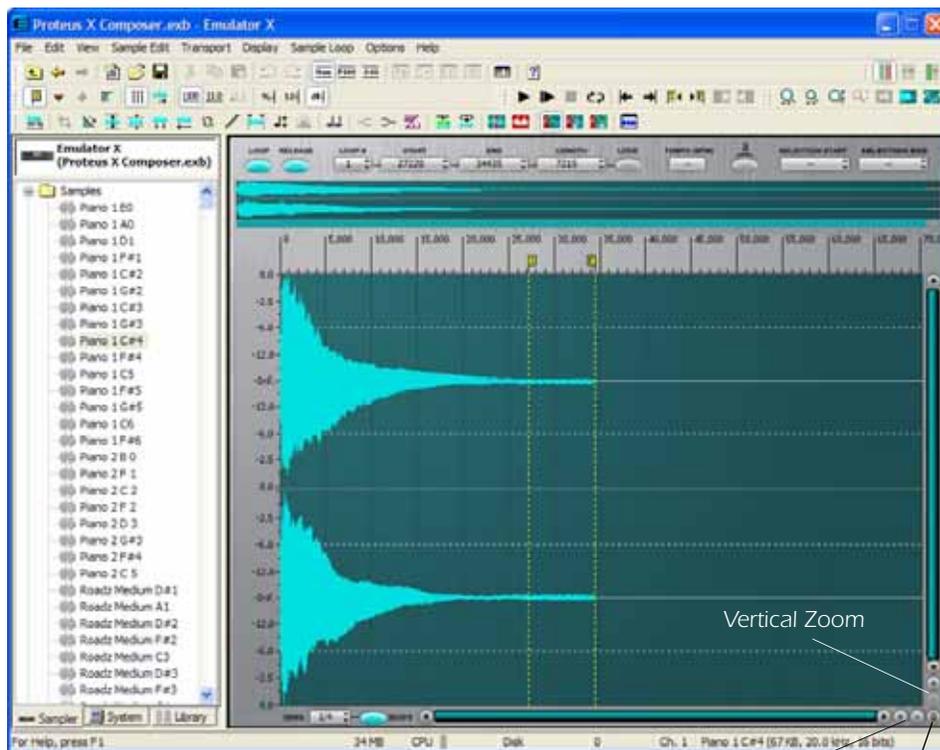
You can also Undo any changes you made by repeatedly clicking the back arrow button (=).

Examine the Samples

Samples are the digital recordings which are the basis of an Emulator X sound. As you've seen, samples can be layered and switched or treated by the Voice Processing functions to create entirely new sounds, but the actual samples can also be modified and sculpted.

A single preset may use over a hundred samples and a bank may contain thousands of individual samples. Let's take a look.

1. Click on the **plus sign (+)** next to the **Samples** folder to open the folder and display the samples in the bank. (If the Preset folder is already open, you can click on the **minus sign (-)** next to the folder to close it and clean up the tree.)
2. Click on any **sample** to display its waveform.



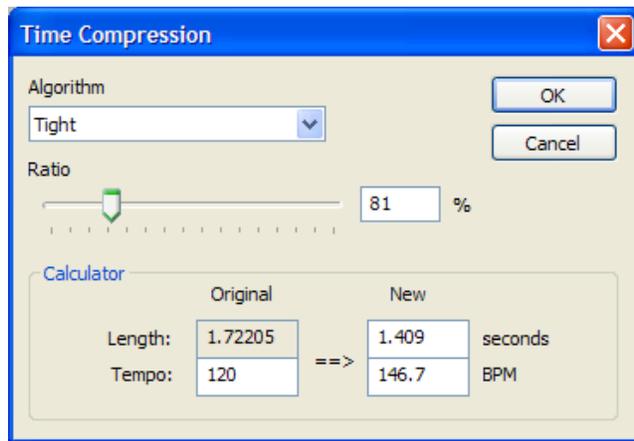
3. The transport controls are located in the toolbar. If you don't see them, right-click an empty area in the toolbar and select **Sample Transport**. Click the **Play** button ► to listen to the sample.
4. Press the **Stop** button ■ to stop playing.
5. Click on the larger magnifying glass of the **Horizontal Zoom** buttons. The time-base of the waveform is magnified allowing you to see more detail. Continue clicking and eventually you'll reach maximum magnification. The individual sample measurements are shown here as dots. If the sample rate is 44.1 kHz, there will be 44,100 of these dots per second of sound.
6. Click repeatedly on the smaller magnifying glass and the horizontal magnification decreases.
7. Click the **Zoom All** button, and the view zooms to show the entire sample.
8. The **Vertical Zoom** buttons allow you to zoom in or out on the vertical axis. None of the zoom buttons change the wave in any way. They're just display controls.

Using DSP

There are a number of DSP or digital signal processing functions that you can perform on the raw samples. Samples can be spliced together, merged, time-stretched, reversed and on and on.

Time Compression/Expansion

1. Select sample **Riff Trip Gb**. Press Play on the transport controls to check it out.
2. Suppose you wanted to change the tempo to fit into your song. The time compression/expansion DSP tool can do this for you. Select **Time Compress** from the DSP toolbar.  The following dialog box appears.



3. **Change the ratio.** Ratios below 100% compress or shorten the time. Ratios above 100% expand or lengthen the time. Note that the tempo and length fields change to show the new length and tempo.
4. Press **OK** to change the time scaling. Press **Play** again and note the change in tempo of the sample.

◆ **Note:** The original tempo shown may not reflect the actual tempo of the sample. You have to set this value manually in order to have the new tempo accurately calculated.

Loop Play

5. Turn on **Loop Play**  on the Transport toolbar. Now the sample will continue to loop until you press **Stop**.

Reverse

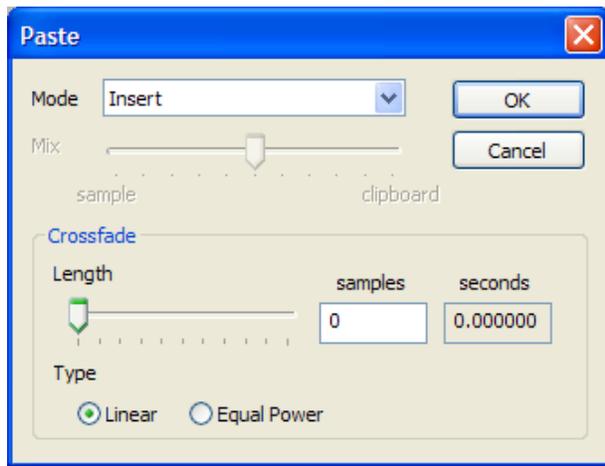
6. Select **Reverse** from the toolbar.  The sample is immediately reversed.
7. Press **Play** and hear the sample played backwards.

Copy and Paste

8. Select the entire sample by pressing the **Select All** button  on the DSP toolbar.
9. Choose **Copy** from the Edit menu. The sample is placed on the clipboard.
10. Click anywhere on the waveform display to deselect the sample, then position the cursor at the very end of the sample by pressing the **Go To End** button  in the transport section of the toolbar. The flashing cursor line is now positioned at the end of the sample.

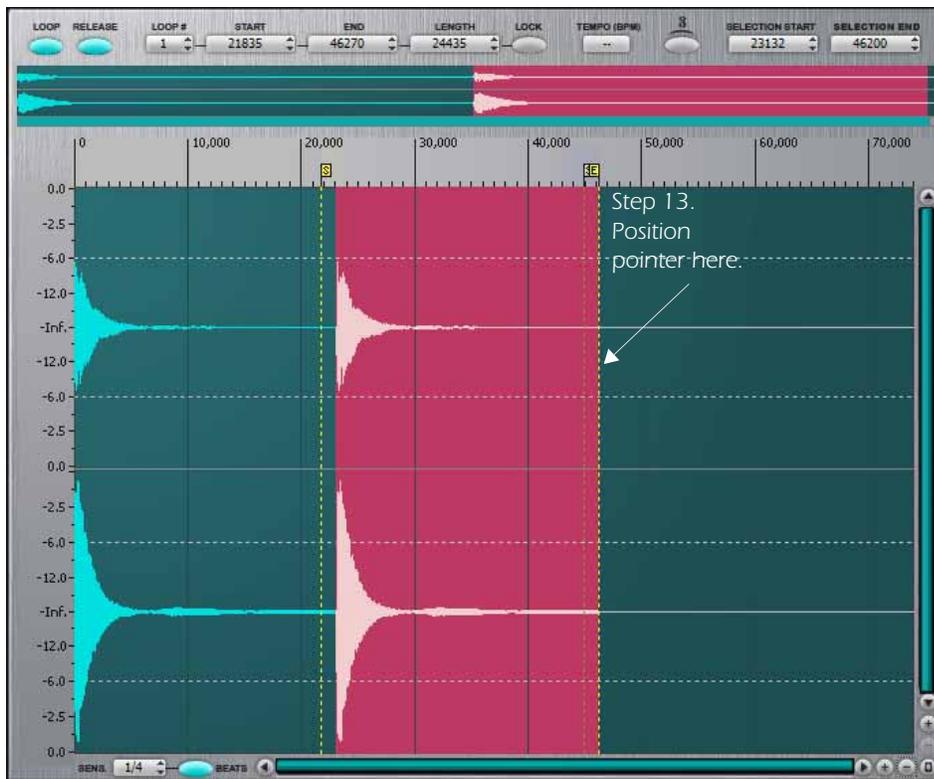
3 - Getting Started
Examine the Samples

11. Choose Paste from the Edit menu. The following dialog box appears.



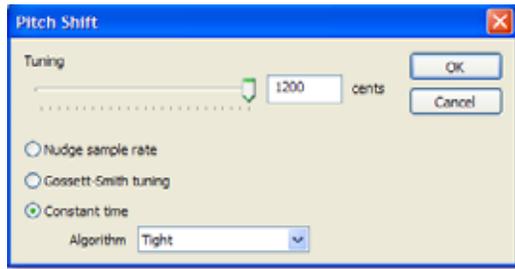
12. Insert mode is already selected and it's the mode we want. The clipboard contents will be inserted at the cursor location, extending the length of the sample. Press OK.

13. Now position the mouse pointer at the very end of the sample. Press the shift key down and click the left mouse button. The section of wave you just pasted is selected.



Pitch Shift

14. Select **Pitch Shift**  from the DSP toolbar. The following dialog box appears.



15. Set the **Tuning** interval. 100 cents equals one semitone. Try a setting of 700 cents or a perfect fifth.
16. Select **Constant Time**. This setting changes the pitch but will leave the rhythm constant. Don't change the algorithm. See [page 192](#) for more information about the pitch change options.
17. Press **OK** to change the pitch.
18. Click on the **Loop Start** button  in the Transport Controls to set the cursor to the beginning of the sample and press **Play**.

Experiment!

Please feel free to try out all the other DSP processes such as Transform Multiplication. Be sure to read the Editing a Sample chapter ([page 169](#)) to learn all about these powerful tools, but you should actually try them out as you read.

Remember that nothing is made permanent until you save the bank, so experiment as much as you like without fear.

Samples can also be exported as WAVE or AIFF files using the **Export** function in the Sample Edit menu. This allows you to use other sound processors in your computer. Externally processed samples can be re-imported using the **Import** option in the File menu.

◆ All the DSP tools are calculated in 32-bit precision math.

Save and Save As

When you choose "Save As" from the file menu, the entire bank—presets, voices and samples—are re-written to the hard disk. This method, although somewhat wasteful of memory, ensures that all your samples remain bound with your bank.

Be sure to rename your bank when using Save As... to save an edited bank, otherwise you may overwrite data you wanted to keep. When you choose **Save**, (instead of Save As) only the edited presets, voices and samples will be written, saving time and memory.

When you save, you might be prompted with a dialog box telling you have unused samples. Choosing to delete the unused samples removes them from your hard drive, saving space and memory. This dialog occurs any time you have samples in your sample pool that are not assigned to any voices.

Emulator X Bank Structure

Banks are comprised of two distinct entities which need to be kept in the same folder on your computer hard disk. The **Sample Pool** contains the raw sample data used. The **Bank** file contains all the voice, preset and multisample data that makes up the Emulator X bank.



Building a Preset from the Ground Up

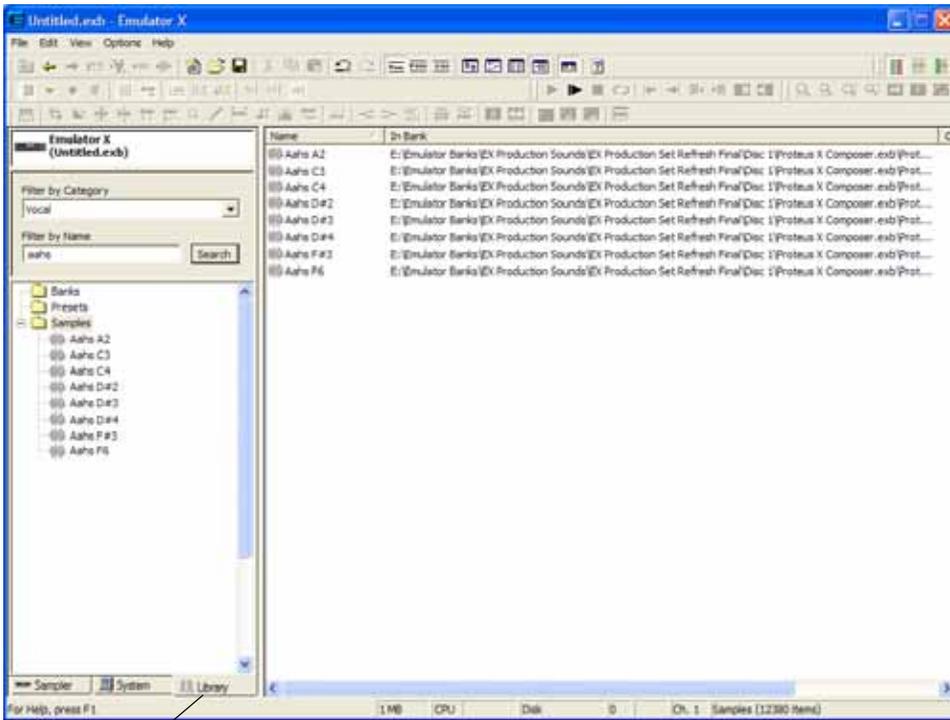
In this tutorial we'll start with an empty bank and create a complete preset using some of the samples in your library. Along the way you'll learn:

- How to locate samples from your library and load them into the bank
- How to place samples on the keyboard
- How to stretch samples across the keyboard
- How to apply various types of audio processing to the samples

► Locate Samples and Load them into the Bank

1. Make sure you have either the Proteus Composer or Proteus X Composer bank on your hard disk.
2. Let's start with a clean slate. Select **New** from the **File** menu.
3. The easiest way to find samples is using the Library feature. Click the **Library** Tab at the bottom of the Tree.
4. Select **Vocal** as the Category, then type "aah" in the 'Filter by Name' box. Click **Search**.

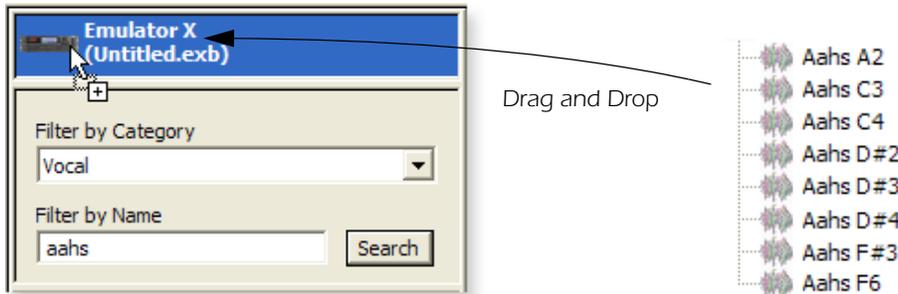
■ **Important:** If the Librarian doesn't find the samples, select **Update Library** from the **File** menu. (You need to have Proteus X Composer bank in your System.)



Library Tab

5. Clicking on the **Samples Folder** in the Tree displays the expanded list in the main window.
6. The window should look something like the one shown above, although you may have more samples shown, depending on the size of your sample library.
7. Locate the following samples: Aahs A2, Aahs C3, Aahs C4, Aahs D#2, Aahs D#3, Aahs D#4, Aahs F#3, Aahs F6.
8. Select the samples listed above and while holding the mouse button, drag and drop them on top of the Emulator X icon above the Tree.

■ **Tip:** Select multiple continuous samples by pressing on the **Shift** key while selecting items. Select multiple non-continuous samples by pressing the **Ctrl** key while selecting items.



9. Click the **Sampler** Tab at the bottom of the Tree, then double-click the **Samples Folder** to open it. You'll notice that the samples have now been added to the Bank.
10. Click on one of the **Sample** icons in the Tree. The Sample Editor appears with one of your Aah waves displayed.



11. The sample is of a fixed length. So that the sample will play for as long as you hold a keyboard key, the sample is looped. Notice (but don't change) the Start and End Loop Markers.
12. Looping is a subject for another tutorial. At this point, let's move on to the next step in building a Preset, and place the samples on the keyboard.

Time to Save?

13. It's time to save whenever you have done enough work that you'd HATE to lose it. Now would probably be a good time.
14. Select **Save As...** from the **FILE** menu, choose a location where you can find it again (the desktop?), and press **Save**. Don't you feel better?

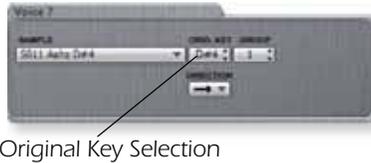
► **Placing the Samples on the Keyboard**

1. Click on the Presets Folder in the Tree, then select **New Preset** (Ctrl+W) from the Presets menu. A new preset appears in the Tree.
2. Select **Voices and Zones** in the new preset. The Voices and Zones page appears.
3. Select **Aahs A2** from the list in the Tree, then **drag and drop** it anywhere in the Key Win area as shown below.

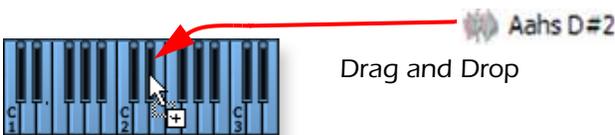


4. A new voice named **Aahs A2** appears in the window. A green bar appears across the entire keyboard range indicating that the sample is transposed across the entire keyboard. The **red bar** shows the original key. Because the original pitch of the sample (A2) is included in the name, Emulator X is smart enough to place the original pitch on the proper key.

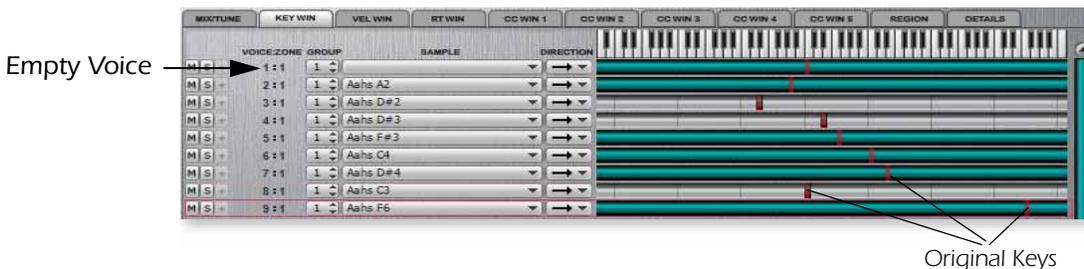
- **Important:** If the original key of a sample is on the wrong key, the sample will be out of tune. The original key selection field is always visible regardless of which tab is selected.



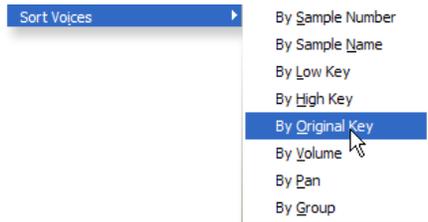
5. Play the keyboard. Since the sample is transposed over the entire keyboard range you can play it polyphonically at any pitch. But as you play in the upper part of the keyboard, notice that the sound doesn't sound so natural anymore. This is a result of the sample transposition. To overcome this problem we add more samples.
6. This time, let's use a different method of assigning samples. This method is useful if the original pitch is NOT included in the sample name. Select **Aahs D#2** from the list in the Tree, then **drag and drop** it over the **D#2** key on the keyboard at the bottom of the window.



7. A new voice named **Aahs D#2** appears in the window on note D#2.
8. Continue placing the rest of the samples. Practice using both methods.
9. Select the **Key Win** tab. This window graphically displays the current sample to keyboard mapping. Depending on the way you placed your samples, the display might look something like this. The **red bars** show the original keys.



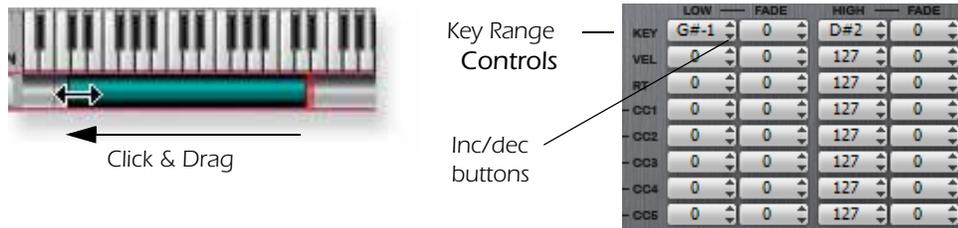
- Notice that the top voice has no sample. This empty voice was created when you created the Preset. We won't need this voice and can get rid of it. Select the empty voice by clicking on it. It becomes highlighted in red. Press the Delete key on your keyboard to delete it.
- Also notice that the original keys of our voices are all over the place. Let's rearrange the voices by the position of the original keys. From the Preset menu, select **Sort Voices, By Original Key**.



- Now the voices should be arranged like the screen below.



- Next, we need to give each sample its own keyboard range. Click on the original key of Aahs D#2 and drag it all the way to the left.



- You can also adjust the key range using the key range controls in the lower right corner of the window. Click the inc/dec buttons and note that the graphic display follows. The Key Range display shows the currently selected voice. The Key Range Controls allow you to select multiple voices and change them all to a single value if so desired.
- Adjust the range of all your voices until the display looks like the one below. Make sure that none of the voices overlap each other and that all the keys are covered.



16. Play the keyboard. The choir sounds much more realistic, especially in the middle range.

Time to Save?

17. It's time to save again. This time just select **Save** (Ctrl+S) from the file menu.

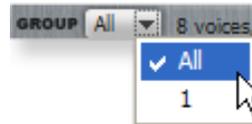
► **Apply Voice Processing**

Now that we have our samples arranged, we can continue to refine our preset using the filters and amplifier.

1. Select the Voice Processing page by clicking the Voice Processing icon in the Tree.

 **Voice Processing** The Voice Processing page appears.

2. In the Emulator X, each voice is a complete synthesizer with its own set of parameters. In most cases, you'll want to treat the voices as a single synthesizer. Select **ALL** voices from the **Group** box above the keyboard in lower part of the window.



- **Important!** Selecting the proper voices to modify is a **fundamental** operation and it's easy to get confused. If the synthesizer controls don't seem to be working correctly, check to make sure you have the correct voices selected.

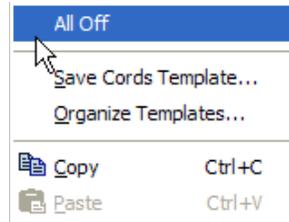


Amplifier Envelope

- Our choir sounds pretty good, but the sound turns on abruptly like an organ. Let's increase the Attack Time a little so that the voices swell up when we press the keys. The Amplifier has its own dedicated Envelope Generator and it's already selected. **Set the Attack 1 knob to about .5 seconds.**
- Much better! Let's increase the Release Time so that the sound dies away slowly when the keys are released. **Set the Release 1 Time to about 1 second.** Heavenly!

Clear the Cords

- Before we begin, take a look at the Cords section for a second. Cords are the virtual wires that connect everything together in the Emulator X. A bunch of Cords are already connected to their default connections. We don't want or need these Cords, so let's get rid of them.
- Right-click anywhere inside the Cords section. A pop-up selection box appears. Select **All Off** to clear all the Cords. Poof!



Phaser Phun

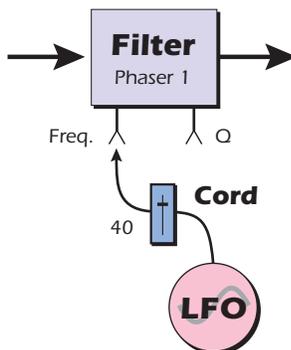
- Now let's add a filter effect. Select **Phaser 1** from the Filter Type menu. This filter simulates a traditional phase shifter by creating two deep notches in the frequency spectrum. Leave the Frequency and Resonance settings at 0.0 for now.
- Let's add a bit of movement to the filter by connecting an LFO to the Filter Frequency. An LFO or **Low Frequency Oscillator** is simply an oscillator that repeats slowly. The LFOs have a range of .08 Hz (12.5 seconds per cycle) to 18 cycles per second.



- Click on the fields to select source and destination for the cord. Set-up Cord 1 as shown at right with **LFO 1+** as the source, **Filter Frequency** as the destination, and an amount of around **+40.00**.



- A block diagram of the connection you just made is shown below.



- Play the keyboard and you'll hear the phaser filter sweeping the two notches across the frequency spectrum.

12. Let's tweak the LFO a bit. Select the LFO/LAGS page by clicking on the LFO/Lags tab. Next, set the Frequency to about 0.9 Hz and change the Shape to Sine.

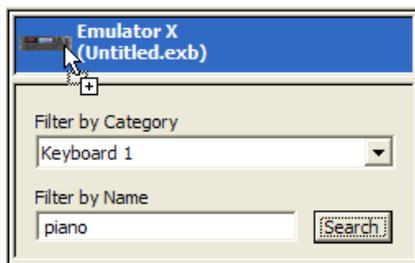


13. If you have trouble adjusting with the knob, try typing the value into the numeric field. One the numeric field "has the focus" you can use the up/down keys on your computer keyboard to increment or decrement values.
14. Now the effect is a lot more subtle. Increase the **Filter Frequency** to about 25 to "zero-in" on the main group of frequency components. Feel free to tune the filter to your own taste. You can also adjust the **Cord Amount** along with the Filter Frequency to get the effect you want.
15. The **Resonance control (Res)** adds resonant peaks and deepens the notches in the filter. Try it out.
16. Explore a little on your own. Go ahead and try out different LFO wave shapes, or different filter types. *(Be careful. Some of the filters have extremely high gain at certain frequencies and they may distort with the harmonically-dense voice samples.)*

► Load an Entire Preset

Now we're going right back to where we started, but instead of loading individual samples, let's load a complete Preset.

1. Select the Library Tab in the Tree.
2. Select **Keyboard 1** as the "Filter Category" and type in piano in the "Filter by Name" box. Click the **Search** button.
3. Double-click on the **Presets** folder in the Tree to open it. You should see several piano presets there.
4. Select one of the piano presets then **drag and drop** it over the Emulator X icon above the filter selections.



5. Switch back to the **Sampler Tab** beneath the Tree and see your piano preset in the bank.
6. Select the piano preset and play the keyboard. That was easy!
7. **Save the bank.**

► **To Audition and Merge Samples**

Samples can also be auditioned and merged.

Locate the Samples Folder

1. Locate the **Samples** folder, just below the Presets folder.
2. Click on the **Emulator X** icon in the Tree to view the Multisetup page.

Audition

3. **Select a Sample** from the Tree and press the **Space Bar** on your computer keyboard. The sample is auditioned from the hard disk.
4. Press the **Space Bar** again to **Stop** auditioning.
5. You can also audition samples by **Right-clicking** over the sample and selecting **Audition**. Right-click and select **Stop Audition** to silence audition.

◆ Samples that reside in .E4B or .EXB disk files can be auditioned from disk.

Merge the Samples

6. **Right-click** over the sample in the Tree and select **Merge** from the menu (or press "M" on your computer keyboard). The sample is added to the current bank.
7. You can also drag and drop a sample over the Emulator X icon to merge it into the bank. These "free" samples will have to be assigned to voices in a preset before they can be played.
8. If you change your mind and don't want to merge the sample after all, choose **Undo** from the Edit menu (or press **Ctrl+Z**) to un-merge the sample.

▼ **Vista Users:** To Audition samples, you may need to download and install the Vista Codec Package 4.5.4.

Signing Off

We've covered a lot in this lesson. Now you know how to create a preset from samples and how to create a custom bank from the presets in the library. This is a very 'deep' instrument, so don't expect to learn it all in one day. Explore a little at a time and most of all don't forget the music, because that's what it's all about. Have fun!

Emulator X Advanced Applications Guide

The Emulator X Advanced Applications Guide contains 43 more in-depth tutorials. The Advanced Application Guide begins with the fundamentals and walks you through nearly every facet of this amazing instrument. You can find this pdf document on your Emulator X3 CD or online at: www.emu.com. Check it out!

The Librarian

As your sample library grows so does the problem of managing all your banks, presets and samples. As you've probably already discovered, a single bank can contain literally thousands of individual samples.

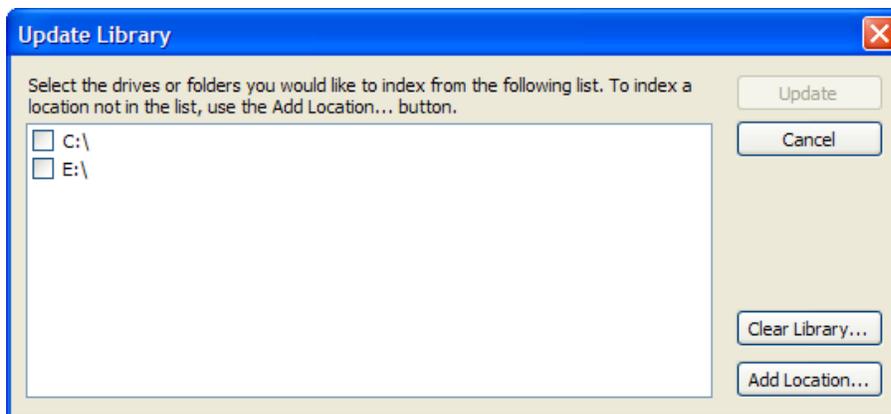
The Library is accessed by clicking the Library tab at the bottom of the Tree. To use the librarian, you first "Update" from the File menu. This creates a searchable database of all your samples, presets, banks and audio files which can be quickly searched. Audio files can be either WAVE or AIFF.

You can select which drives you wish to add to the catalog or manually "Add Locations" to narrow down the selection range to speed the update process. The Add Location button also allows you to specify network drives to catalog. Only the selected targets are overwritten with new data when the update process is performed. Unselected targets remain as they were. The "Clear Library" button erases the entire library so you can start from scratch.

► Using the Librarian

Catalog your Disks

1. Select **Update Library...** from the File menu. A popup dialog box appears asking you to select the disks you wish to catalog.



Update

Updates the Library with the selected items, leaving non-checked targets intact.

Clear Library

Erases Library database.

Add Location

Allows you to manually select targets to speed update time or select network drives to catalog.

2. Select any disk drives that contains the banks, presets and samples that were included when you installed the Emulator X application. Using the **Add Location** button, you can select network drives to add to the Library catalog. The Add Location button also allows you to select specific folders or locations for cataloging.
3. Click the **Update** button to begin cataloging. The display will read: "Updating". This process may take a minute or two depending on how many files need to be cataloged.

■ **Cool Tip:** Categorize multiple Presets:

1. Click the Sampler Tab in the Tree.
2. Click the Presets folder.
3. Select multiple presets.
4. Right-click and choose category.
5. Select category.

Find your Files

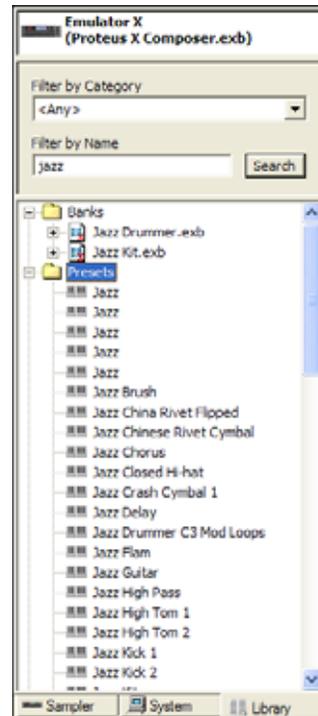
4. **Select the Library tab** at the bottom of the Tree and press **Enter** on your computer keyboard. All Emulator X Banks, Presets, or Samples can now be accessed here. (This can take a few seconds, especially with very large banks.)
5. Click on the **Presets** folder to view all the presets on your hard disks.
6. Selecting an item in the **Category** field returns any Banks, Presets or Samples in the selected category.
7. Type a search string in the **Filter by Name** field, then press the **Search** button to search the library.

The 'Filter by Name' field is a subset of the Category search. Select **<Any>** in the Category field to search the entire library.

8. Browse through the Bank, Preset, and Sample folders to find the sounds you want.
9. **Right-click on a Preset or Sample** to Audition the Preset or Sample from disk. You must play the keyboard while Auditioning to hear presets.

Load your Files

10. To add a file to the bank, **Left-click and Drag** the desired object in the Library over the Emulator X icon at the top of the tree and release the mouse button. The file is added to the bank.
11. **Or...** Select the object you want from the Library and **Right-click**. Then select **Merge** to merge the object into the current bank.



4 - Multisetup / Preset Globals

The Multisetup

The Multisetup is the top level of the Emulator hierarchy and is the place where you choose and assign presets to each of the MIDI channels. All the control settings in this window are saved when you save the Multisetup.

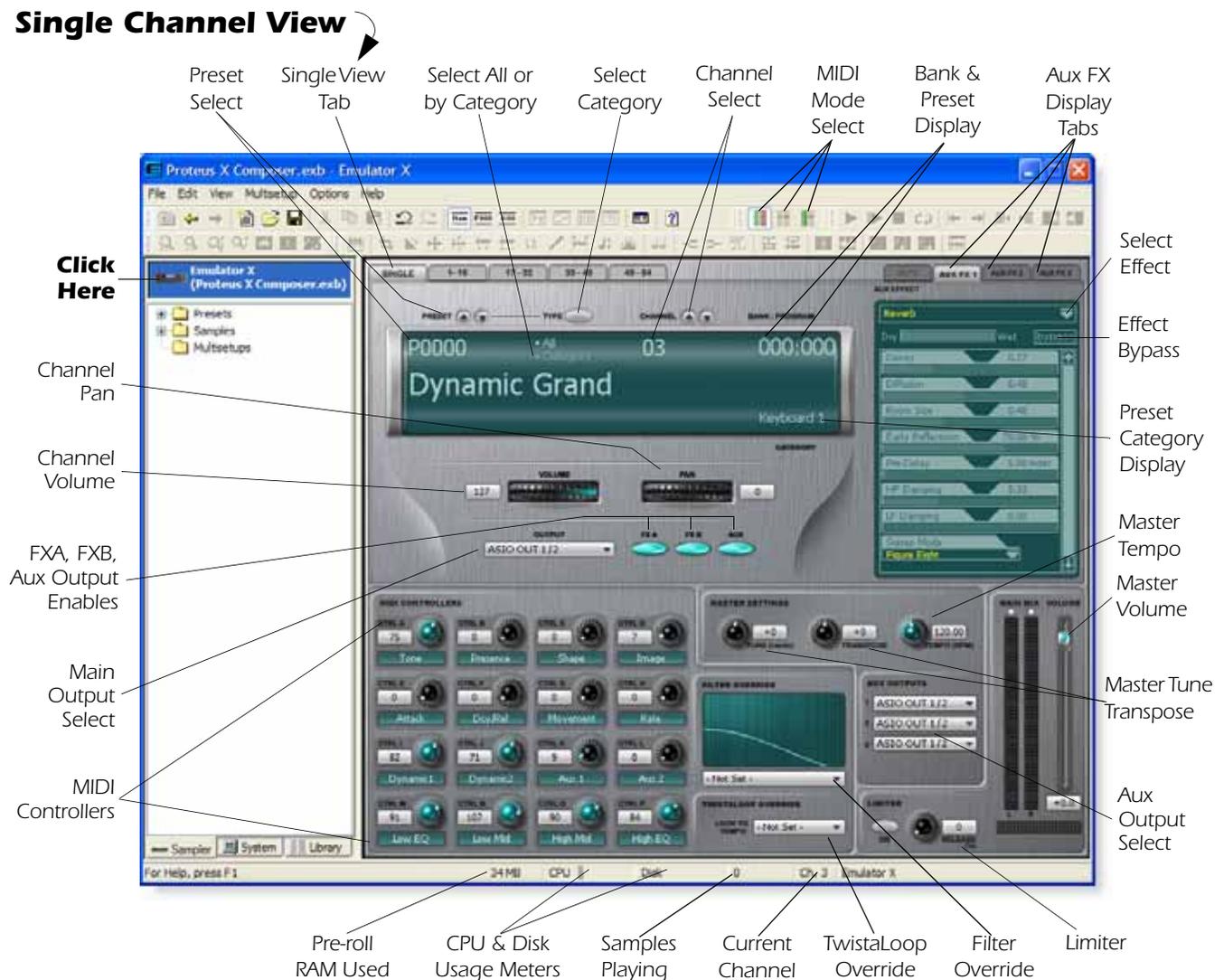
A multisetup assigns a preset, volume, pan position and output routing to each of the 64 MIDI channels. Multisetups can be saved and recalled for use with a particular song or sequence so that all the MIDI channels play the proper preset.

▼ The MIDI Controller settings and Filter Override are not stored with the Multisetup. These settings are stored with each Preset.

► To Select the Current Multisetup:

1. Click on the Emulator X icon at the top of the Tree. The current multisetup appears.
2. The Single Mode view is shown below. This view is convenient for performing live or when you're only playing one preset at a time.

Single Channel View



There are five views of the Multisetup page. The **Multisetup View** buttons change the current view. The **Single** channel view is shown below.

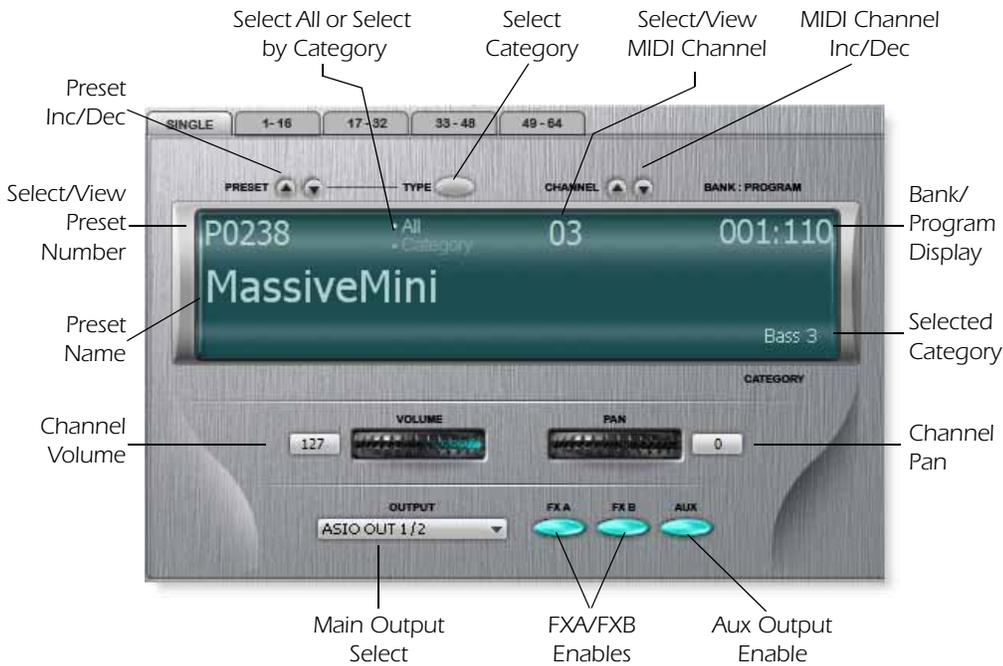
- **Single**..... Displays the preset for the currently selected MIDI channel.
- **Channels 1-16**..... Displays the preset, volume and pan for MIDI channels 1-16.
- **Channels 17-32**.... Displays the preset, volume and pan for MIDI channels 17-32.
- **Channels 33-48**.... Displays the preset, volume and pan for MIDI channels 33-48.
- **Channels 49-64**.... Displays the preset, volume and pan for MIDI channels 49-64.

Changing Presets (from Single View)

Single View is the default view when you first run Emulator X. This view is convenient for performing live or when you're only playing one preset at a time. All the main parameters for the selected MIDI channel are displayed in Single View and a convenient Category mode allows you to quickly select presets in a certain category. A close-up of the current channel settings is shown below.

◆ Note that Single View only changes the display and does not affect the currently selected MIDI mode (i.e. Omni, Poly or Multi).

Single View Close-up



◆ The Bank/Program Display shows the Bank and Program number you would use to select the preset from a sequencer.

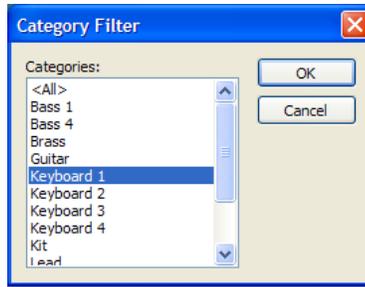
► To Select Presets by Number

1. Click on the **Preset Increment/Decrement** buttons to move up or down the presets by number.
2. Highlight the **Preset Number** field and type in the desired preset number.

■ **Tip:** You can also select a preset by dragging and dropping it from the tree into the single view screen.

► **To Select Presets by Category**

1. Click on the Select Category button. The pop-up dialog box at right appears.
2. Select the desired preset category, then press **OK**.
3. Choose **Select By Category** from the Single view screen above.
4. Use the **Increment** or **Decrement** buttons to select presets. Note that only presets of the selected category are selected.



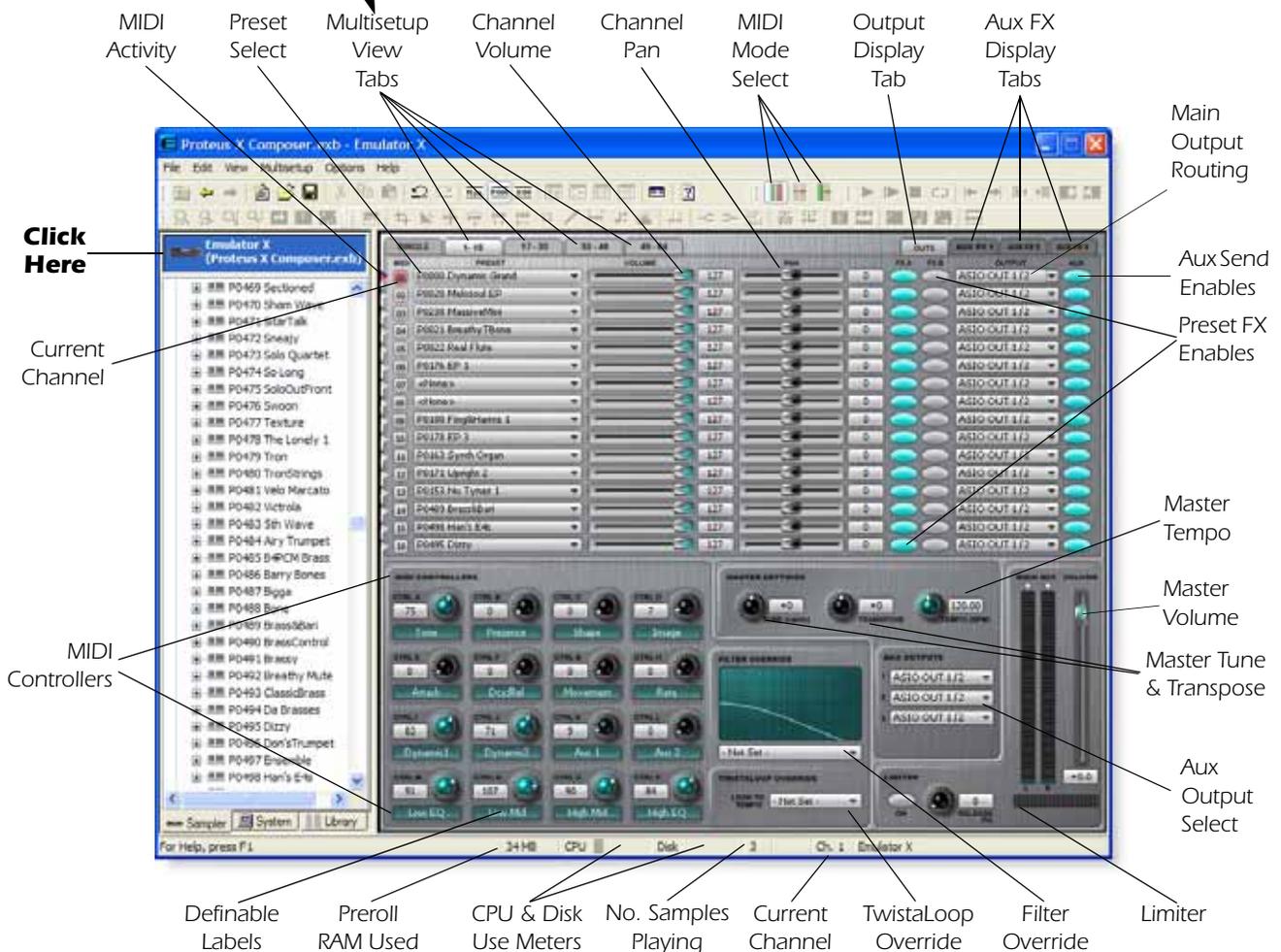
► **Changing the Current MIDI Channel**

The Current MIDI Channel is simply the MIDI channel you are currently working with. To change the current MIDI channel, simply increment or decrement the channel number using the buttons, or click on the channel number and type in the desired number.

Multi Channel View

Press the 1-16 Multisetup View button to view 16 channels at once. This mode is useful for sequencing or when playing multiple MIDI channels.

Multi Channel View



The Current MIDI Channel

Change the current channel by clicking anywhere on the desired horizontal channel strip. The red indicator displays the currently selected channel number. The MIDI Controllers change to show the settings of the currently selected MIDI channel.

► To Select a Preset:

1. Click on the **Preset Select** button for the desired MIDI channel. The Select Preset dialog box appears.
2. Click on the presets in the list to audition them. Presets can be viewed by large icon, small icon, by list, or by detailed list. Presets can also be sorted by name, ID, bank/program number, or category when in Detail View mode.
3. To choose a preset, select it and press **OK**, or **double-click on the preset** in the list.

► To Select a Preset from the Tree:

1. Set the Current Channel by clicking anywhere on a horizontal channel strip in the Multisetup window. The channel number turns red, indicating that this is the current channel.
2. Choose the preset you want in the tree and **Right-Click** on its preset icon.
3. Choose **Select on Current Channel**. The preset is now selected for that channel.
4. You can also select a preset for the current channel by simply clicking the preset's keyboard icon from the tree.

MIDI Channel

Each channel only responds to MIDI data on its particular channel. Use the floating keyboard (select **Keyboard** from the **View** menu) to audition presets on the currently selected channel. Select the current channel by clicking anywhere on the desired channel strip. The red indicator displays the currently selected channel number.

Preset Assignment

Assign presets to MIDI channels by clicking on the triangle selector button ▼ to the right of the preset field. A pop-up dialog box appears with the list of presets in the bank. Select a preset, then click **OK** to select.

Channel Volume

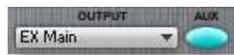
This control sets the relative volume of the MIDI channel. This is equivalent to MIDI controller #7 (channel volume) and changes made over MIDI will be shown here.

Channel Pan

This control sets the stereo position of the MIDI channel. This is equivalent to MIDI controller #10 (channel pan) and changes made over MIDI will be shown here.

Output Assignment

This control selects the **Main Output** assignment for the MIDI channel. 100% of the stereo output signal is sent to the selected output bus.



The number of outputs available in this field depends on the number of stereo channels your sound card provides. Add ASIO input strips in PatchMix DSP to increase the number of available outputs. Output destinations will be labeled according to the PatchMix DSP input strip scribble strip name. The EX session templates in PatchMix DSP are designed to be used with Emulator X.

▼ **Important:** The act of creating PatchMix DSP ASIO mixer strips makes them available for Emulator X. If they are not created in PatchMix DSP, you won't see them in the Output Assignment field.

Output View Button

This button allows you to view the Aux Send On/Off, Main Output Routing, and Preset FXA/FXB Bypass buttons when not in Single Mode. (The button is disabled in Single mode.)

You may want to bypass the Preset Effects on certain channels in order to free up CPU power or to hear the presets dry.

Aux FX View Buttons

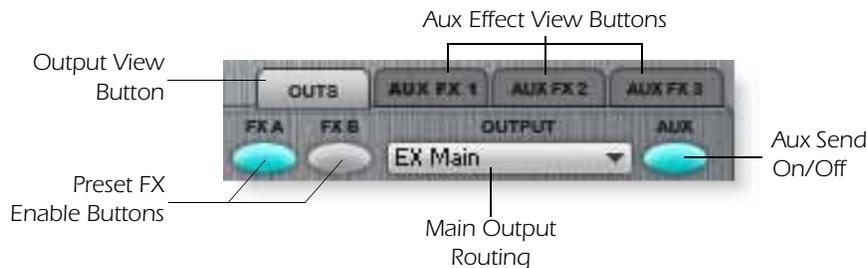
The buttons above the TV display select which of the three possible Aux Effect is currently displayed. Note: The Output View button is disabled in Single mode.

Aux Send On/Off

These buttons enable (blue) or disable (grey) the inputs to the Aux Sends for each MIDI channel. When a factory FX template (such as Proteus X Composer) is used, these buttons serve to enable/disable the Aux FX for each channel. The Main Output is unaffected by these buttons.

These buttons disable the Aux Sends whether the Aux FX are being used or not. Please refer to the diagram on [page 234](#) to see the exact location of this switch in the signal path.

◆ The Output View button is only available when any of the multimode views are selected. In Single mode you can only control the current channel.



Preset FX Enables

These two rows of buttons marked FXA and FXB, enable or disable the preset effects for each MIDI channel. These work differently than the Aux Send On/Off buttons in that they *bypass* the effects instead of turning them on or off. These buttons are useful to bypass any preset effects which may have been programmed in the preset.

Please refer to [Chapter “9 - Effects” on page 231](#) for detailed information about the output and effects routing.

Select Effect

Click on the Effect Select icon  at the top right corner of the Effects TV screen. A list of available effects drops down. See [“Adding an Aux Effect in the Multisetup” on page 232](#) for complete details on how to add aux effects.

TwistaLoop Override

This three-way switch determines if the beat looping features are locked to Master Tempo. The three modes are as follows:

- **On:** Beats on the channel are locked to the Global Master Tempo. This control overrides the “Sync To Master Clock” setting in the preset.
- **Off:** Beats on the channel are NOT locked to the Global Master Tempo. This control overrides the “Sync To Master Clock” setting in the preset.
- **Not Set:** The “Sync To Master Clock” button in the Preset Globals page determines whether or not the Beats are locked to the Global Master Tempo.



Global Controls



Master Volume Control & VU Meter

This is the master volume control for Emulator X. This slider controls the output volume of all MIDI channels. The range of this control can be adjusted using the Headroom/Boost control in the preferences dialog box. [See page 24.](#)

The Master volume control responds to the Universal Real Time System Exclusive message for Master Volume. If your MIDI controller (e.g. an E-MU Xboard) sends such messages (such as from a master volume slider), the master volume in Emulator X changes accordingly.

The VU meter indicates the overall output level of the Emulator X Main Outputs. It serves as a handy indicator to show clipping or to troubleshoot audio output problems.

Tempo Control

This control and associated display sets the master tempo when Emulator X is chosen as the tempo source. Emulator X contains a global master clock which is used for Tempo-based Envelopes, Tempo-based LFOs, and clock modulation.

“External Tempo Source” in the MIDI Preferences dialog must be Off for this control to operate.

When “External Tempo Source” in the preferences dialog box is On, and an external MIDI device or application is generating MIDI clock, the Emulator X tempo will lock to the MIDI clock tempo.

Master Tune

Master Tune adjusts the overall tuning of all presets so you can tune Emulator X to other instruments. The master tuning range is ± 1 semitone in one cent steps (1/100th of a semitone). A master tune setting of 0¢ indicates that Emulator X is precisely tuned to concert pitch (A=440 Hz).

Master Transpose

The master transpose parameter transposes the key of all presets in semitone intervals. The transpose range is ± 12 semi tones (1 octave).

Limiters

When playing Emulator X you may sometimes hear digital clipping. This clipping occurs in the digital hardware and is not caused by Emulator X, which uses high resolution floating point processing and has tremendous internal headroom.

To avoid the clipping, you could always lower the volume controls, but there’s a better way. The Limiter monitors the signal ahead of the output stage and automatically turns down the volume before clipping can occur.

► **To Use the Limiter**

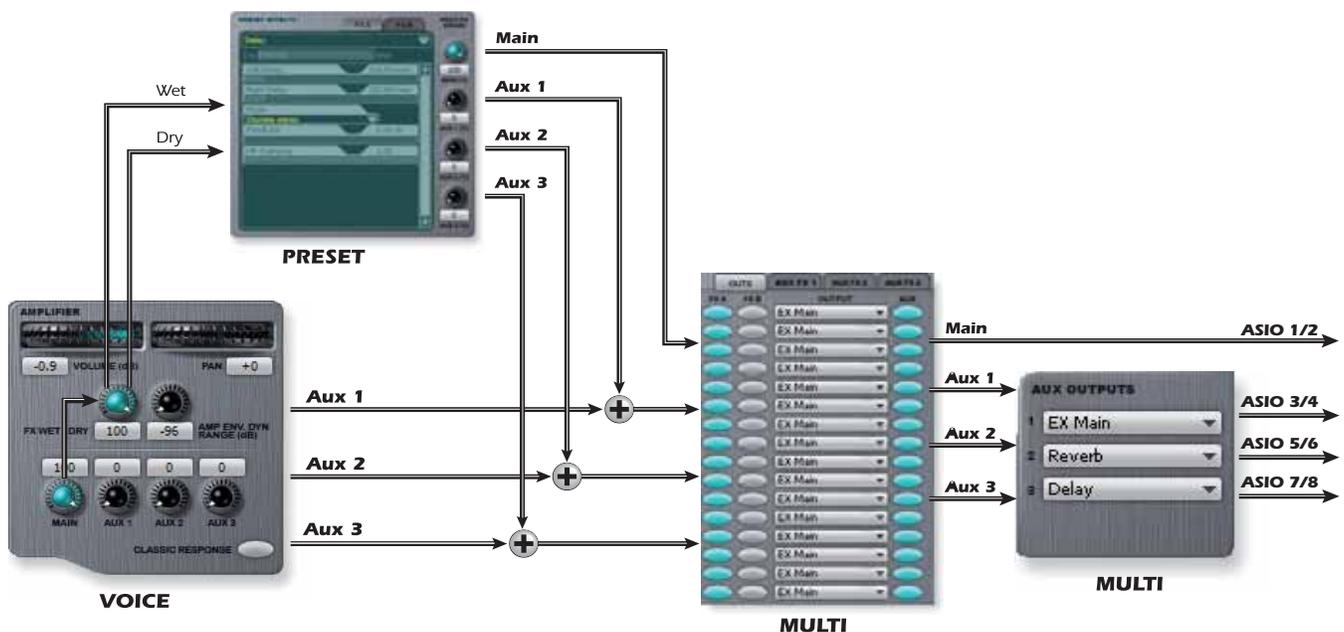
1. Press the Enable button, illuminating the button.
2. Set the Release knob to about 30% to start. Release controls how fast the limiter turns the volume back up after reducing it to avoid the clipping peaks.
3. Increase the release time if you hear obvious volume changes (pumping). Reduce the release time for percussive playing.

Aux Outputs

These three output assignments assign the Aux Routings selected in the Voices or Presets to pairs of ASIO channels or to DirectSound. Every Voice (page 36) has three effects sends with programmable amounts as well as a wet/dry mix to the Preset FX section. The three effect sends from all voices and Presets are summed into the three Aux Outputs where they are assigned to an ASIO pair. The Aux ASIO streams can be routed to the PatchMix DSP mixer or to other software devices running on your computer.

◆ See Chapter “9 - Effects” on page 231 for complete descriptions of the effects routing.

Emulator X Aux Bus Routing



Using the Emulator Sessions in PatchMix DSP

The E-MU Digital Audio System contains special Sessions that can be used with the Emulator X factory banks. These Sessions connect the Aux Outputs to PatchMix DSP mixer input strips containing PatchMix DSP hardware effects.

If a factory bank has the Aux Outputs set as shown in the diagram above, it is designed to use the PatchMix hardware effects. If all the Aux Outputs are set to EX Main it most likely uses the built-in software effects. You can easily add PatchMix DSP hardware effects to a bank that uses Emulator X software effects by simply switching an Aux bus to a PatchMix strip with effects.

See “Block Diagram of the Effects Routing” on page 234.

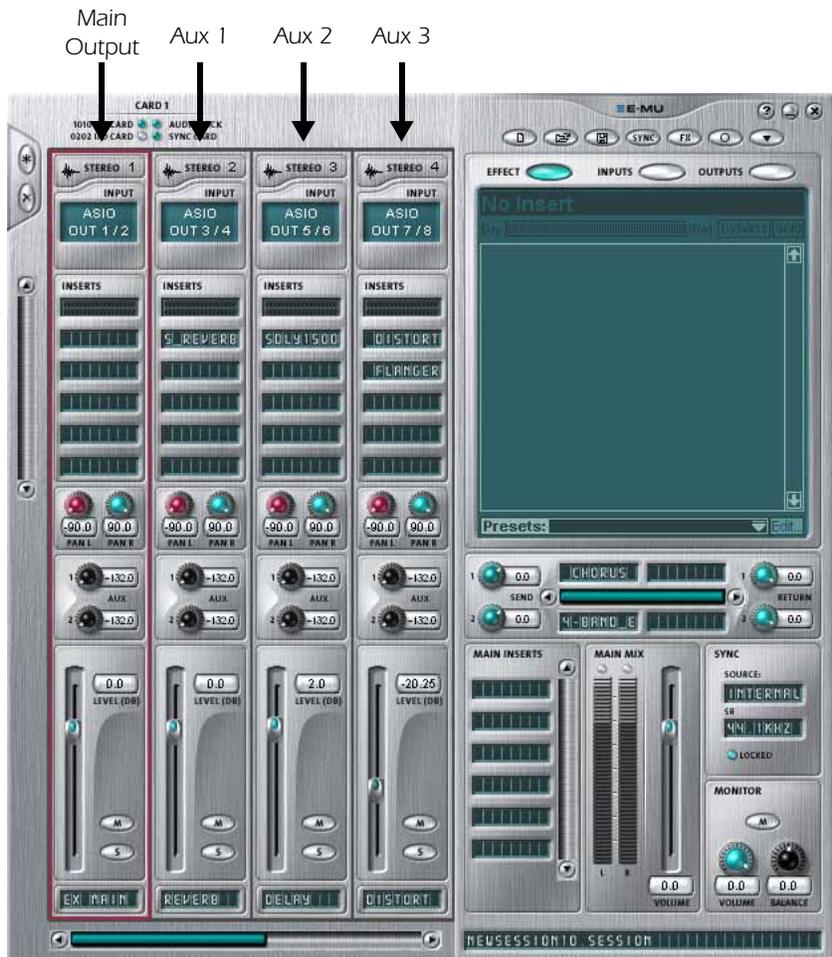
IMPORTANT

The act of creating PatchMix DSP ASIO mixer strips makes them available for Emulator X or any other application. You must create the ASIO mixer strips before Emulator X or any other application will see them.

► To Open the Emulator Sessions in PatchMix DSP

1. Open PatchMix DSP by clicking on the  icon and select New Session.
2. Select Emulator X Studio.
3. Select Percussion EQ for use with the percussion banks on Disk 3. Select Emulator X Studio for all other banks.
4. The PatchMix DSP screen below shows the Main and Aux Output connections from Emulator X.

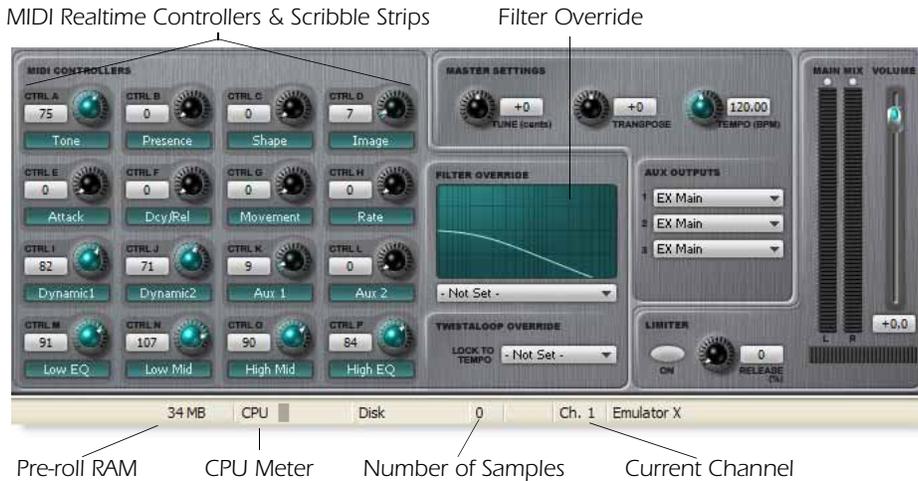
Emulator X Studio Session



The routing options in the Emulator X Aux Output section take their names from the PatchMix DSP scribble strips. To route an Aux Send to a hardware effect, simply change the Aux Output routing to a PatchMix strip containing effects.

Filter Override

The filter type of the first voice in the preset (on the currently selected channel) is shown in this display. If you change the filter type in this field, ALL the voices in the preset will change to the filter type you select. When the display reads, - Not Set - the filters programmed in each voice will be used. The filters may be set differently for each voice.



MIDI Controllers & Scribble Strips

Sixteen realtime MIDI Controllers are available to control the preset on the currently selected channel. These controllers are connected to MIDI sources as assigned in the Preferences dialog box ([page 28](#)) and changes made over MIDI will be reflected here.

The scribble strip names and the initial settings of the controller knobs are user programmable in the Preset Globals window. See [page 80](#).

Pre-roll RAM Used

This display shows how much of your computer's RAM is being used by pre-roll sound data. The amount of Pre-roll RAM can be adjusted in the preferences dialog box. When Sample Streaming is disabled this number shows the total RAM usage. See [page 21](#).

CPU Meter

This display shows how much of your computer CPU power is being used by Emulator X. If the meter turns red, this indicates that there is not enough CPU bandwidth to properly process the audio in a timely manner, which may result in audible glitches or delayed notes.

Number of Samples Playing

This display shows you how many samples are currently being played. The maximum number of samples that can be played depends on a variety of factors such as: HD speed, RAM memory speed and CPU speed. See ["Audio Setup" on page 23](#) for more information about performance.

Current Channel

This displays the currently selected MIDI channel. Change the current channel by clicking anywhere on the desired channel strip.

Working with Multisetups

A Multisetup is a “snapshot” of the current Emulator X settings. Multisetups store the preset, volume, pan position, and output bus routings for 64 MIDI channels along with the Global Tempo and the three Aux Send destinations.

Saving FX & Multisetups

It is important to realize what data is stored in a multisetup when saving banks. In addition to Volume, Pan, and Preset choices for each channel, the Initial Controllers, FX Aux 1, Aux 2, and Aux 3 are stored with the multisetup. The send level for each preset to each Aux bus is stored in the preset data. **Remember to save the bank whenever you have made changes that you want to keep.**

VSTi Multisetups

A multisetup for your VSTi is stored in your Cubase project and will be loaded when the project is opened. This multisetup overrides whatever default multisetup may have been saved if the bank was opened in standalone mode. If you wish to store extra multisetups you may do by using the Multisetup menu on the toolbar.

► To Store the Current Multisetup

1. Set up all the multisetup parameters the way you want them.
2. Select **Store** from the Multisetup window on the toolbar.
3. Choose a **name and number** for the multisetup. Changing the number allows you to place the Multisetup anywhere you want in the list.
4. Press **OK** to store the Multisetup.
5. **Save the Bank**, otherwise nothing will be saved.

► To Restore a Multisetup

1. Open the multisetup folder in the Tree. Any multisetups contained in the bank will be listed below the folder.
2. **Left-click** on the desired multisetup icon in the tree and **drag it** over the **Emulator X icon** above the tree. This multisetup will be restored.
3. Alternate Method - **Right-click** over the Multisetup icon in the Tree and choose **Load**.

► To Export the Multisetup as a New Bank

This feature lets you save only the presets and samples you need as a new bank.

1. Set up all the multisetup parameters the way you want them.
2. Select **Export** from the Multisetup window on the toolbar. The popup “Save As” dialog box appears prompting you for the save location.
3. Choose the location for your new bank and click **Save** to save the bank, or click **Cancel** to cancel the operation.

► To Rename a Multisetup

1. **Open** the multisetup folder in the Tree. Any multisetups contained in the bank will be listed below the folder.
2. **Right-click** on the desired multisetup icon in the tree, then select **Rename** from the pop-up menu.
3. Type in a new name for the multisetup.

► **To Delete a Multisetup**

1. Open the multisetup folder in the Tree. Any multisetups contained in the bank will be listed below the folder.
2. **Right-click** on the desired multisetup icon in the tree, then select **Delete** from the pop-up menu.
3. The Multisetup will be deleted.

► **To Duplicate a Multisetup**

1. Open the multisetup folder in the Tree. Any multisetups contained in the bank will be listed below the folder.
2. **Right-click** on the desired multisetup icon in the tree, then select **Duplicate** from the pop-up menu.
3. The Multisetup will be duplicated renamed "Copy of...XX" and placed in the first empty Multisetup location.

► **To Rearrange the Order of Multisetups**

1. Click once on the multisetup folder in the tree. The list of multisetups appears.
2. Select **Details View** from the view options in the toolbar and change the ID number.
3. If you try to save a Multisetup with a number that is already used, a popup dialog box will appear asking you if you want to use the next available ID number or enter a new ID.

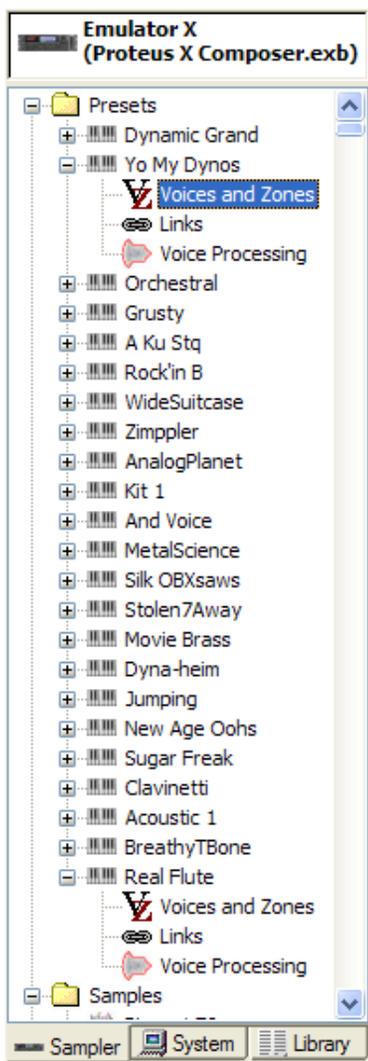
Preset Globals

The Preset Globals allow you to edit presets and their key maps. Samples can be arranged into single-sample or multi-sample voices with synthesizer processing applied. Then these complete voices can then be arranged anywhere on the keyboard.

The Preset Editor contains three sections. The first section, **Voices & Zones**, is used for arranging voices and samples on the keyboard and editing global preset parameters.

The **Links** section makes it easy to build new presets by combining other finished presets. Multiple presets can be crossfaded or cross-switched according to key position, key velocity, or by the position of MIDI continuous controllers.

The third section, **Voice Processing**, is used to perform more detailed editing of voices such as envelope shaping of amplitude and filters as well as modulation parameters. This is the synthesizer section of Emulator X containing E-MU's legendary Z-plane filters and powerful matrix modulation.



► To Edit a Preset:

1. Select the **Sampler** tab from the Tree. The contents of the current Emulator X bank appears in the Tree.
2. **Double-click** on the **Preset** folder from the tree (or *click once on the (+) symbol*). The preset folder expands to display the presets contained within.
3. Click on one of the **Preset** icons in the Presets folder to view the Preset Globals window.
4. **Double-click** on the **desired preset icon** from the tree (or *click once on the (+) symbol*). The preset icon expands to display the subsections: **Voices and Zones**, **Links**, and **Voice Processing** levels.
5. Select the desired section: **Voices & Zones**, **Links**, or **Voice Processing** by clicking on the associated icon.

► To Select a Preset from the Tree:

1. Set the Current Channel to the one you're using by **clicking on the channel number** in the Multisetup screen.
2. Choose the preset you want in the tree and **Right-Click** on its preset icon.
3. Choose **Select on Current Channel**. The preset is now selected for that channel.

These parameters affect the entire preset and are saved with the bank.



- You can quickly access the Preset Globals window by clicking the  button in the toolbar.

Transpose & Volume

Transpose works by shifting the keyboard position. The actual samples are not returned. In other words, if you have the same sound on each key, such as a piano, the piano will change key when you transpose. On the other hand, if you have a different drum assigned to each key, changing transpose will shift the position of the drums on the keyboard.

The volume control allows you to adjust the volume for the entire preset. This control is variable from -96 dB (off) to +10 dB.

Initial Controllers A-P

These set the initial value of MIDI controllers A-P for the preset. As soon as a MIDI controller is moved, it jumps immediately from this initial setting to the new setting of the control. The controllers A-P are assigned to MIDI continuous controller numbers in the Preferences dialog box.

Controllers A-P can be adjusted via MIDI or by turning the controller knob in the Multisetup.

The Label fields to the left of the initial controller knobs are user definable. These labels or "Scribble Strips" also appear in the Multisetup screen to show the function of the realtime controller knobs.

See ["MIDI Channels & Real-time Controls" on page 314](#) for additional information about Controllers A-P.

- Changing the initial controller settings is an easy way to customize a preset. Simply change the knobs and save the bank.

Capture MIDI Controllers

You can capture the state of all your MIDI input device's MIDI controllers, and place the settings in the Initial Controllers section of the Preset. Essentially, this allows you to capture the Multisetup's MIDI controller state, and store it in the preset.

► To Capture MIDI Controllers

1. Make any desired changes to any MIDI controllers on your MIDI input device, or in the Multisetup window.
2. Go to the Preset Globals window for the preset you want to apply your current MIDI state.
3. From the Preset menu, select **Capture MIDI Controllers**.
4. Notice that the initial controllers now reflect the state of your MIDI controllers. To keep these settings, you can save the Preset.

TwistaLoop Lock To Tempo

This control locks all analyzed beats in the preset to the global master tempo. When this function is on, all beats in the preset will lock to the master tempo unless "TwistaLoop Override" ([page 73](#)) is turned off.

This button works in combination with "TwistaLoop Override." When TwistaLoop Override is "Not Set," this button determines whether beats sync to the master tempo for the preset.

When TwistaLoop Override is set to On or Off for the MIDI channel, this button has no effect. These two controls allow you to set the state of "Sync to Master Clock" by MIDI channel (TwistaLoop Override) or by Preset (this button).



Initial Tempo

This feature allows the master tempo to be programmed from presets that use tempo-based envelopes, LFOs or TwistaLoop.

The Initial Tempo works in combination with the "Preset Tempo Channel" option in the Preferences menu, which allows one specific MIDI channel to control the master tempo. [See Preset Tempo Channel on page 27.](#)

When the Initial Tempo is set to a tempo on the defined Preset Tempo Channel, the master tempo will follow this tempo setting when the preset is selected.



► To Set the Master Tempo from a Preset:

1. Turn **Preset Tempo Channel** on and set it to the desired MIDI control channel. (Options, Preferences, MIDI) *"External Tempo Source" must be turned off.*
2. Set the **Initial Tempo** (Preset Globals) to the desired tempo.
3. Turn on **TwistaLoop - Lock To Tempo** (Preset Globals) if you are using TwistaLoop.
4. Select the preset on the assigned preset tempo channel.

Tuning Tables

In addition to the standard equally divided octave tuning, Emulator X contains twelve factory programmed tuning tables. The Keyboard Tuning parameter selects which tuning is used by the current preset.

The factory keyboard tuning tables are described in the following table.

Tuning Tables	Description
Equal Temperament	Standard Western tuning 12 equally spaced notes per octave.
Just C	Just intonation. Based on small interval ratios. Sweet and pure, non-beating intervals.
Just C2	Allows you to play the following chords in the key of C: C, E, F, G, A, B, C#m, D#m, Em, F#m, G#m, Am, Bm
Just C-minor	Allows you to play the following chords in the key of C: C, E, F, G, A, B, Em, Am, Bm, C#m, D#m, G#m
Just C3	Allows you to play the following chords in the key of C: C, D, F, Bb, C#m, Dm, Em, F#m, G#m, Am
Vallotti and Young	Valotti & Young non-equal temperament. Similar to 12 tone equal temperament. Each key has a different character for a given scale.
19-Tone	19 tone equal temperament. 19 notes per octave. Difficult to play, but works well with a sequencer.
Gamelan	5 tone Slendro and 7 tone Pelog. (Javanese.) Pelog are white keys, Slendro are black keys. Exotic tunings of Gamelan flavor.
Werkmeister	A "well" temperament developed in the 17th century. Although you can play in all keys, each key sounds slightly different.
Kirnberger	Another well temperament developed by Johann Philipp Kirnberger where no pitch is more than 12 cents off from equal temperament.
Scarlatti	A variant of Meantone tuning which was used from the 15th to 18th centuries.
Repeating Octave	Middle C octave is repeated up and down the keyboard. Link with a preset in equal temperament to form unusual inversions up and down the keyboard.
al-Farabi 12	A twelve-note reduction of an Arabic/Middle Eastern tuning system. The following scales are useful: Diatonic: C Eb G G# A A# B c Zalzal: C Eb Gb G# A A# B c Persian: C Eb F G# A A# B c
22-Tone	22 tone equal temperament. Good for minor harmonies.
24-Tone	24 tone equal temperament. These notes are quarter steps apart.
36-Tone	26 tone equal temperament. Difficult to play, but even better for minor harmonies.

The Just C Tuning Tables

Well Tempered and Just were standard keyboard tunings up until the 20th-century when the current “equal tempered” scale became prevalent. In an equal tempered scale, the octave is equally divided into 12 parts. In Just or Well Tempered scales, the 12 notes are separately tuned to produce pure chords. However, in Just tunings you are limited to playing certain chords and if you play the wrong chord it may sound sour!

Tuning tables can be changed as you play using a program change (create several presets with the same sound and different tuning tables) or using a continuous controller (link 2 presets and crossfade between them using a controller).



The Just C2, Just C min, Just C3 Tuning Tables

Fully explaining the mysteries of just intonation is beyond the scope of this manual, but the subject is covered exhaustively in Hermann Helmholtz’s [On the Sensations of Tone](#), available at most libraries and bookstores.

The four just intonation tables are called Just C2, Just C3, and Just C Minor. Try playing in the key of C/Cm using each table. You’ll quickly discover both the wonders and the frustrations of just intonation! In Just C, for example, you’ll find that the chords C, Em, F, G, and Am sound beautiful. If you hold one of these chords, you’ll hear no “beating”. After playing for a few minutes, switch back to Equal Temperament for a rude awakening!

At this point you might wonder why anyone would use Equal Temperament to begin with. For the answer to that question, play a D, Dmi, or Bb chord! The intervallic ratios that make the C & G chords sound so pure make the D chord sound horribly out of tune. That’s why we had to include Just C3. In this tuning, D, Dmi and Bb sound in tune, but the G chord will sound wrong.

Each of the 4 tables allows you to play a different group of common chords in just intonation. Sadly, there is no single 12 note tuning that will allow all of the common chords to be in tune, and of course that’s why they invented the equal temperament tuning system that we use today.

Just C

Play these chords: C, E, F, G, A, Cm, C#m, Em, F#m, Gm, Am

Just C2

Play these chords: C, E, F, G, A, B, C#m, D#m, Em, G#m, Am, Bm

Just C minor

Play these chords: C, D^b, D, E^b, G, A^b, Cm, Em, Fm, Gm

Just C3

Play these chords: C, D, F, B^b, C#m, Dm, Em, F#m, G#m, Am

Equal Temperament Tunings

For each of the equal temperament tunings except 19-tone, middle C (i.e., MIDI note 60) is middle C as it would be in a standard 12-note tuning. The keyboard is then scaled up and down from middle C (C2). For example, using the 22-tone scale, one octave above middle C is 22 keys up the keyboard (A#3).

Key CC Window

These controls work in conjunction with the “Keyboard - CC Window” function in the Voices and Zones. [See “Continuous Controller Window Pages” on page 108.](#)

Pressing the “Key CC Win” button and setting a Keyboard Range fields marks off a section of the keyboard to be used as control keys for the CC Window or a PatchCord source in the Voice Processing section. These keys will no longer trigger sounds.

Instead, they are now used for selecting different voices in the CC Windows, or as a modulation source in the Cords. ([See page 222.](#))

Preset Modulators

The Preset Modulators are special class of modulation sources in that they *originate* at the *Preset Level*, but their output is used in the *Voice PatchCords*. There are three types of preset modulation sources.

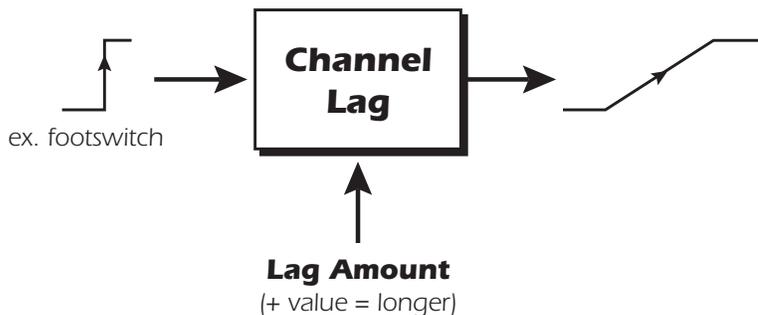
Preset Modulator	Typical Uses
Channel Lags 1 & 2	“Leslie” speaker effects. Switch slowly increases/decreases LFO speed.
Poly Ramp (Timer)	Controlling key-up layer volume based on the time a key is held.
Channel Ramp	Ramps on First Note only. “Hammond” style organ percussion.

Channel Lag 1 & 2 Rate

Like the Layer Lag processors (described on [page 138](#)) the Channel Lag slows down rapid changes in the input signal. The output “lags” behind the input at a pre-programmed rate.

Unlike the layer level lag processors, the *Channel Lags 1 & 2* take effect as soon as the preset is selected. In contrast, the Voice level lag processors begin acting only after a keyboard key has been depressed. The Channel Lag also has a *Lag Amount* input which controls the lag time. Positive lag amounts increase the lag time. A MIDI controller (*front panel knob*) is commonly used to control lag amount.

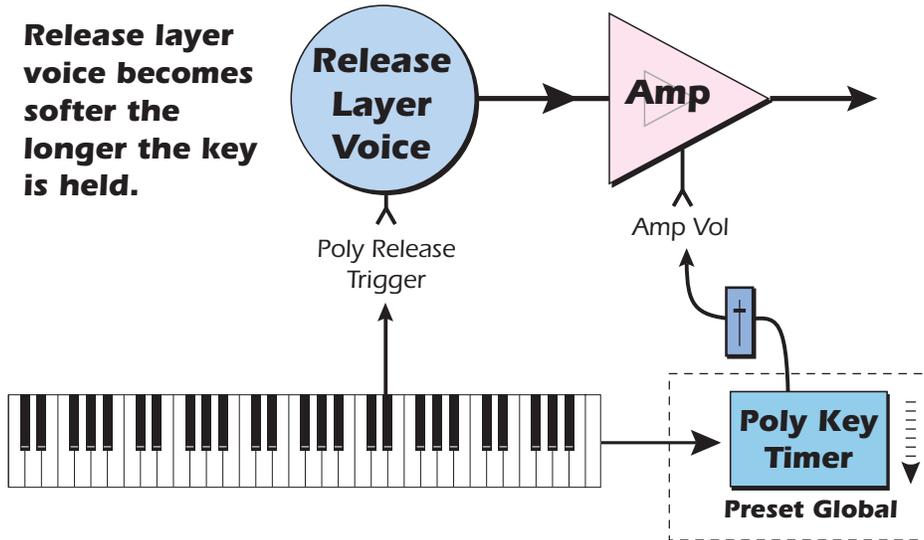
The Channel Lags could be used to slowly increase or decrease the speed of an LFO which in turn controls some other effect, perhaps left/right panning, pitch, or the filter. When a MIDI footswitch is used as the input, the lag acts to slow down the instantaneous change of the switch. The slowly changing output value can then be routed using a patchcord to smoothly crossfade between voices or gradually change the speed of an LFO.



Poly Key Timer

This processor starts a timer which begins counting down whenever a key is pressed, and stops and outputs its value when the key is released. The value of this timer can be used on any voices after the key release. The Poly Key Timer is a special modulation source controlled by the key and not the voice.

The Poly Key Timer is commonly used on grand piano presets to reduce the pedal damper volume the longer the keys are held.



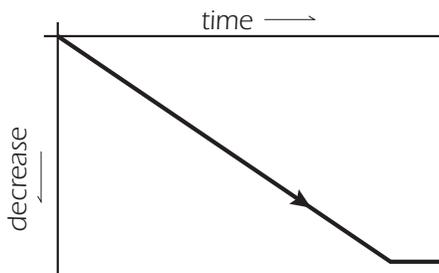
Note-On - Poly Key Timer begins counting down.

Note-Off - Poly Key Timer stops counting and outputs value.

Channel Ramp

This processor generates a single negative going slope whenever the first key is depressed on the MIDI channel. This preset level processor was originally designed to simulate the percussion key click on Hammond organs, but you may find other uses.

Channel Ramp



1st Note Played

The Channel Ramp instantly resets when all notes have been released and the next *first key* on the MIDI channel is pressed. The value of the ramp rate sets the increment. Large values produce fast ramps and small values produce slow ramps.

To use the Channel Ramp as a first-note decay envelope, patch the Channel Ramp out to Amp Volume and set the Amp Envelope for a fast attack and 100% sustain and short release.

Preset PatchCords

Preset PatchCords give you real-time control of the Preset Modulators. There are 16 Preset PatchCords per preset with a source, a destination and an amount control. The amount can be set from -100 to +100.

Modulation Sources	Modulation Destinations
Off	Off
Pitch Wheel	Channel Lag 1 In
Mod Wheel (Modulation Wheel)	Channel Lag 1 Amount
Pressure (Channel Pressure)	Channel Lag 2 In
Pedal	Channel Lag 2 Amount
MIDI Volume (Controller 7)	Channel Ramp Rate
MIDI Pan (Controller 10)	Post-FX Main Send
Expression (Controller 11)	Post-FX Aux 1 Send
MIDI A-P	Post-FX Aux 2 Send
Footswitch 1	Post-FX Aux 3 Send
Flip-Flop Footswitch 1 (Fswitch 1 FF)	FX A Mod 1
Footswitch 2	FX A Mod 2
Flip-Flop Footswitch 2 (Fswitch 2 FF)	FX A Mod 3
Thumbswitch	FX A Mod 4
Flip-Flop Thumbswitch (Tswitch FF)	FX B Mod 1
DC Offset	FX B Mod 2
Channel Lag 1 & 2	FX B Mod 3
Channel Ramp	FX B Mod 4

Templates

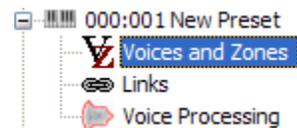
For each different module in the Preset Globals window, you can save a library of your favorite settings. These Templates make it easy to create your own customized voices and presets. Templates are saved as an XML file within the Emulator X application directory so they're always available for use. For more information on templates, [See Templates on page 279.](#)

5 - Voices & Sample Zones

Each preset contains voices. A voice houses a sample, or multiple samples, and can cover an individual key, or a range of keys.

The Voices and Zones window allows you full control over the parameters for each voice, including levels, tuning, keyboard range, crossfading, looping, region, and more. The Voices and Zones window is divided into tabs, each with its own set of controls. Certain controls, such as Volume and Pan appear in each window.

To access the Voices and Zones window, expand a preset in the tree, and click **Voices and Zones**.



The screenshot shows the 'Voices and Zones' window for a preset named '000:001 New Preset'. The window is divided into several sections:

- Page Selection Tabs:** MIX/TUNE, KEY WIN, VEL WIN, RT WIN, CC WIN 1, CC WIN 2, CC WIN 3, CC WIN 4, CC WIN 5, REGION, DETAILS.
- Table:** A table with columns for VOICEZONE GROUP, SAMPLE, DIRECTION, VOLUME, PAN, X'POSE, COARSE, and FINE. It lists 14 voices with their respective sample names and parameters.
- Overview Section:** Located at the bottom, it includes controls for MUTE, VOLUME, PAN, FINE TUNE, COARSE TUNE, and TRANSPOSE. It also shows a keyboard diagram with a highlighted key range (C3 to C4) and a MIDI controller range.
- Labels:** Various callouts point to specific features: 'Selected Voice/Zone' points to the first row; 'Expand/Collapse Multisample' points to the '+' icon; 'Mute/Solo Voice or Zone' points to the 'M S' buttons; 'Selected Voice(s)' points to the '4' in the 'VOICEZONE GROUP' column; 'Overview Section' points to the bottom control area; 'Key Range for Selected Voice(s)' and 'MIDI Controller Range' point to the keyboard diagram.

Voices and Zones Pages

The Voices and Zones Window is separated into tabbed pages:

Mix/Tune

Adjust volume, pan, transpose, coarse tuning, and fine tuning for each voice. For more information, see ["Mix/Tune Page" on page 99](#).

Key Win

Edit keyboard range assignments and place voices and multisamples wherever you want. For more information, see ["Key Window Page" on page 99](#).

Vel Win

Control the volume of a voice by key velocity. Crossfade or cross-switch between voices by key velocity. For more information, see ["Velocity Window Page" on page 103](#).

RT Win

Control the volume of a voice using a real-time controller such as a modulation wheel, a pedal, an LFO or an envelope generator. For more information, see ["Realtime Window Page" on page 105](#).

CC Win 1-5

Switch CC windows using specified keys on your MIDI controller. For more information, see ["Continuous Controller Window Pages" on page 108](#).

Region

Define which part of a sample plays. For more information, see ["Region Page" on page 115](#).

Details

An overview of each voice that allows you to view and change Sample, Direction, Mix/Tune, and Key Window information using numeric values. For more information, see ["Details Page" on page 119](#).

- You can quickly access the Voices and Zones window for the current preset by clicking the  button in the toolbar.

Voice Selection

In each tabbed page, voices can be selected by clicking any part of the voice strip. Selection is shown by a red box around the voice strip.

1. Select the first voice in the “Voices & Zones” screen.
2. Hold the **Shift** key to select contiguous voices. Hold the **Control** key to select or deselect non-contiguous voices. Press **Control-A** to select all voices.

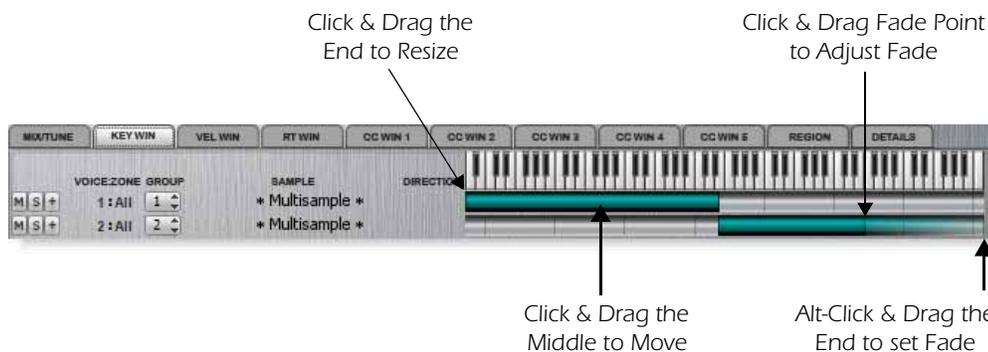
Manipulating Voices

Under each tabbed page, there are various controls that allow you to change a voice’s Group, Sample, Direction, Volume, Pan, etc. Some controls, such as Group or Direction, are selected from a drop-down menu. Others, such as Volume and Pan, are controlled by a slider or wheel.

Most pages also contain parameters that can be adjusted using graphical sliders.



You edit a slider’s range or position by clicking the slider, and dragging your mouse left or right. Create a fade by holding the **Alt** key in conjunction with clicking and dragging your mouse. See the example below:



- To set the high and low ranges, click and drag the end points of the graphic display
- To move the whole range without resizing it, click and drag the middle of the range.
- To create a fade, hold **Alt**, and click and drag the end points. Once you create a fade point, you can click and drag the fade point to re-position it.

Working With Multiple Voices

If you select more than one voice, you can edit parameters for all selected voices simultaneously. To select more than one voice, hold the **Ctrl** key while clicking the voices with your mouse. To select contiguous voices, hold the **Shift** key while clicking.

It’s important to understand how changing parameters in the various tabbed pages differs from changing parameters in the Overview section that occupies the bottom half of the screen:

- In the tabbed pages, such as the Mix/Tune page, parameter changes work in a *relative* manner. This means if you select multiple voices and decrease the Volume of a voice by 10 dB, you decrease all voices’ volumes by 10 dB.
- However, if you select multiple voices and change the volume knob in the Overview section, you change all selected voices’ *absolute* volumes.

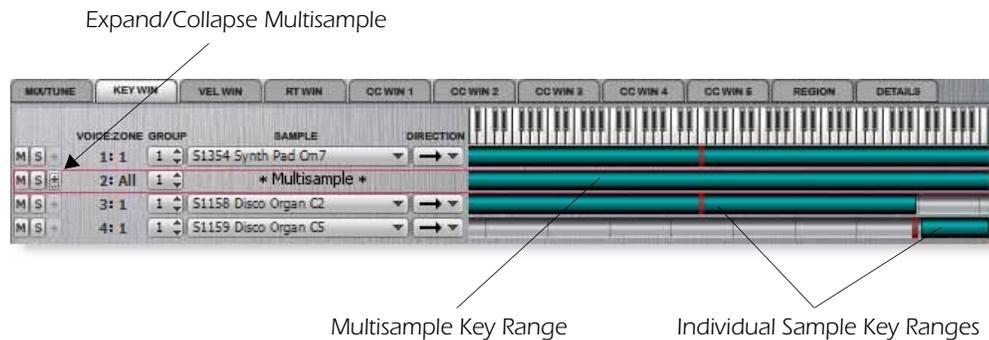
Note about multiple voice editing: If you open several voices for editing that have different voice parameter settings, the parameters in the different voices will remain as they were until you change them. That is, if you have three voices selected and you change the filter cutoff, all three voices will jump to the new cutoff setting, but all the other parameters will remain as they were.

Samples & Multisamples

The samples contained in the voice are displayed and can be edited. If the voice contains more than one sample, the word "Multisample" is displayed in the sample field. All the samples in a multisample share the same set of synthesizer parameters. However, the volume, fine tune and pan parameters can be adjusted separately for each sample in a multisample.

► To View & Edit the Samples in a Multisample

1. Click the + symbol to display the individual samples and key ranges which comprise the multisample.
2. Click the - symbol to collapse the multisample.



Multisamples allow you to place multiple samples under the same set of synthesizer voice parameters.

The multisample key range sets the overall key range limit for the samples in the multisample. In the diagram above, the top sample won't be played, because its key range is outside the limit of the Multisample key range.

Groups

Voices can be categorized into Groups as a way to organize and select similar voices, say a layer of piano voices and a layer of string voices, into unified entities.

You can create a temporary group if you want to edit several voices. Grouping the voices and then selecting the group is faster and more dependable than manually selecting each voice when editing.

► To Place a Voice into a Group

1. Tag each voice you want to be in the group with the same group number. This can be done in the “Voices and Zones” window or from the “Voice Processing” window.
2. Whenever you select that group number in the Voice Processing window, all voices with that group number will be selected. You can define up to 99 groups, but a voice can only be in one group at a time.

► To Make a New Group from the Currently Selected Voices

1. Select the voices you want to be in the new Group from the “Voices & Zones” page.
2. Select **New Group** from the Preset menu. A new group will be created using the lowest available group number.

► To Select All Voices for Editing

- From the Voices & Sample Zones window, press **Control-A**.
- From the Voice Processing window, select **All** in the Group box.

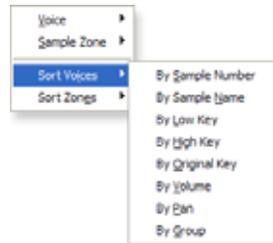
► To Select a Group of Voices for Editing

- Method #1: Simply select the desired group number from the Voice Processing page. Once you have selected a group, you will be editing all the voices in this group.
- Method #2: Select any voice in the group, then select **Select Group “N”** from the **Preset** menu. All the voices in the group will be selected.

Sorting Voices and Zones

You can sort Voices, and Zones using the following criteria:

- Sample Number
- Sample Name
- Low Key
- High Key
- Original Key
- Volume
- Pan
- Group (voices only)



► To Sort Voices or Zones

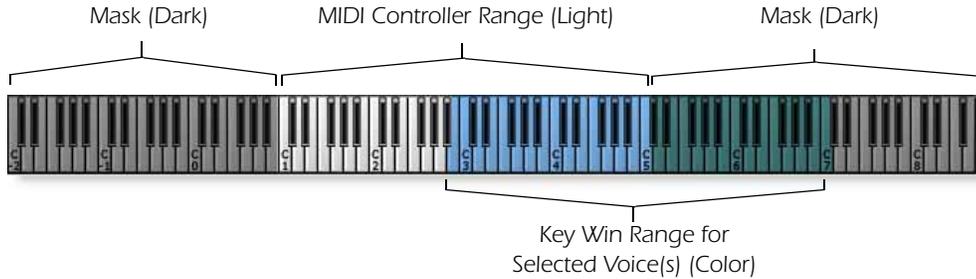
1. Right click any Voice or Zone number.
2. Select **Sort Voices** or **Sort Zones**, and choose how you want to sort them.

The Keyboard Display

The keyboard display on the bottom of the screen gives you a visual representation of the Key Win range for the currently selected voice(s), and the Note range of your MIDI controller.

Notes that fall within the currently selected voices' key range appear in color.

You can also mask out notes that don't fall within a range of keys that match your MIDI controller. Masked keys display in darker colors.



► To Choose What type of MIDI Controller to Display

- **Right-click the keyboard** and select the type of controller you are using. Choices are Full MIDI (no mask), Piano (88 keys), Three Octave, XBoard 61, XBoard 49, and XBoard 25.

The keyboard display is also useful for identifying notes and creating a new voice with a specific Original Key value Key Win range:

- When you hover your mouse over the keyboard, the name of the key is displayed.
- You can create new voices by dragging samples onto the keyboard in the Voices and Zones screen. See [“Adding Samples and Voices” on page 97](#) for more information.

◆ The Keyboard also appears in the Voice Processing screen.

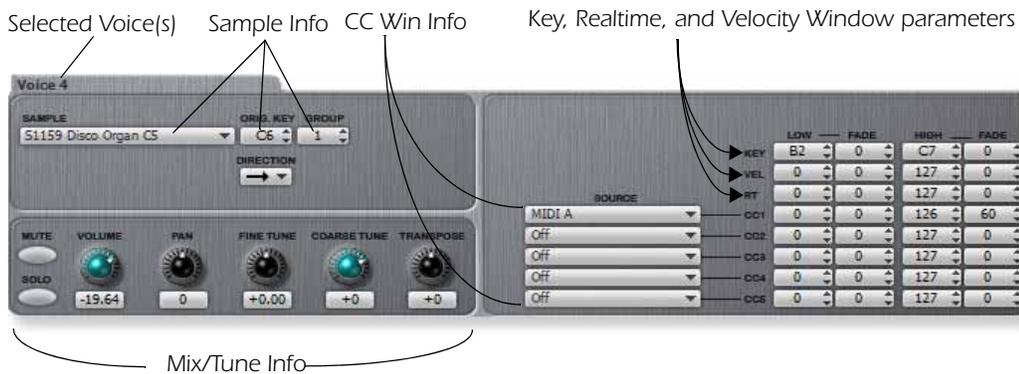
Overview Section

The bottom portion of the page contains an overview that displays and allows you to control parameters for the selected voice(s).

These controls are available regardless of which tabbed page you select (except the Region page), and work per voice. When you modify a setting here, the corresponding setting changes in the Voice Processing window.

Note that the selected voices are indicated in the upper-left corner of the Overview. Changes to these controls only affect the indicated voice(s). If you select multiple voices and change a parameter in the Overview section, that parameter changes for all selected voices.

When you select multiple voices, it's important to understand how changing parameters in the Overview differs from changing parameters in the various tabbed pages. In the Overview section, all changes are *absolute* values. This means if you select multiple voices and change the Volume knob to a value of 10, you change all selected voices to a volume of 10. However, if you select multiple voices and move the volume slider in the Mix/Tune page, you change all selected voices' *relative* volumes.



■ **Tip:** To set a control to its default setting, hold **Alt** while clicking the control, or right-click the control, and select **Default**.

Sample Controls



Sample

The sample assigned to the selected voice or zone. If you select multiple voices, this window appears blank.

Original Key

Sets the Original key of the selected voice or zone.

Group

Changes the group of the selected voice or zone.

Direction

Selects whether the sample plays forward, backward, forward loop, backward loop, or one-shot.

◆ Muting and Soloing voices does not save polyphony! Muted voices still play; they are simply silenced.

Mix Tune Controls

These are the same controls found in the Mix/Tune page.



Mute

Mutes the selected voice(s) or zone(s).

Solo

Mutes all voices but the selected voice(s) or zone(s).

Volume

Controls the volume for the selected voice(s) or zone(s). This control is also located in the Voice Processing window.

Pan

Controls the Pan for the selected voice(s) or zone(s). This control is also located in the Voice Processing window.

Fine Tune

Fine tunes the pitch of the selected voice(s) or zone(s) by cents. This control is also located in the Voice Processing window.

Coarse Tune

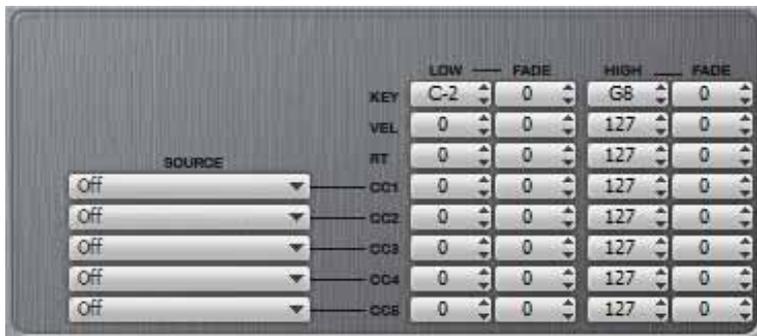
Tunes the pitch of the selected voice(s) by semi tones. This control is also located in the Voice Processing window. This control is NOT available for individual zones of multi-sample voices.

Transpose

Transposes the pitch of the selected voice(s) by semi tones. This control is also located in the Voice Processing window. This control is NOT available for individual zones of multisample voices.

■ **Tip:** To quickly remove all solos and mutes, select **Clear All Solos and Mutes** from the **Preset** menu.

Key Win, Velocity Win, RT Win, and CC Win Controls



Rather than using the graphical representation found in each corresponding tabbed page, you can change the Key Win, Velocity Win, Realtime Win, and CC Win values from any page (except the Region page). Remember, any changes you make only affect the selected voice(s).

◆ See [page 129](#) to understand the difference between transpose and coarse tune. You can also set the group number from this page.

Direction

For each voice (or zone in the case of a multisample voice), you can choose direction settings. The **Direction** drop-down menu allows you to choose one of the following:

- Forward**  The region plays from beginning to end while the note is held.
- Backward**  The region plays in reverse from end to beginning while the note is held.
- Forward Loop**  The region loops forwards from beginning to end as long as the note is held.
- Backward Loop**  The region loops backwards from end to beginning as long as the note is held.
- One-Shot**  The region plays from beginning to end when a note on signal is received. If you release the note before the sample completes, the region still finishes playback.
Note: You may have to adjust the Amp Envelope settings in order to hear the sample. See “Envelope Complete” mode on [page 126](#).

Adding Samples and Voices

You can create new voices and add samples to them from any of the tabbed pages in the Voices and Zones window.

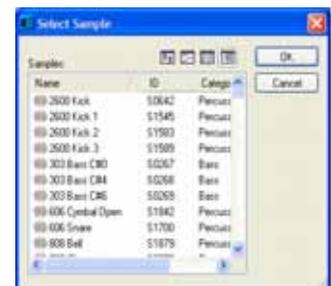
► To Add a New Voice

There are several ways to add a new voice from the Voices and Zones window:

- Press **Ctrl+W** on your keyboard.
- From the **Preset** menu, select **New Voice**.
- Right-click an empty area and select **New Voice**.

► To Change a Zone’s Sample

- From the **Sample** column, click the Selector Button ▼ next to the appropriate voice or zone. Select a sample from the Select **Sample** dialog box.
- Drag a Sample from the Tree to the zone’s strip.



► To Add an Additional Sample to a Voice

This either adds to, or creates a Multisample voice. From the Voices and Zones window:

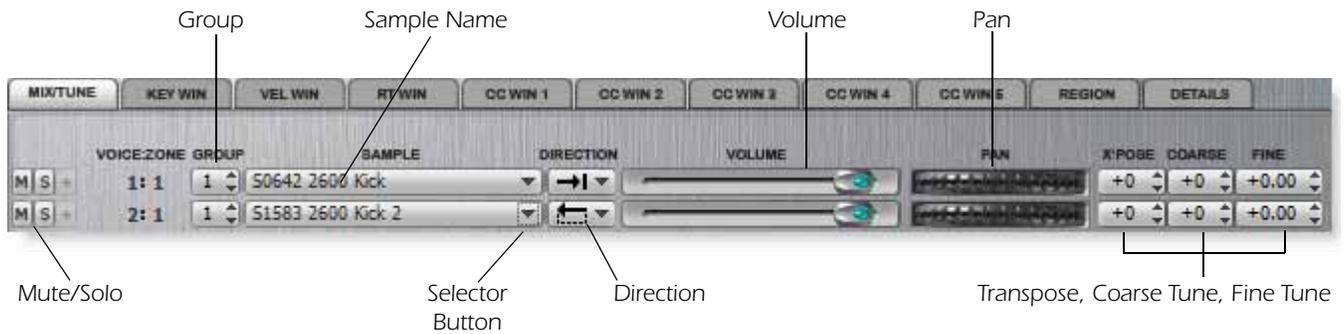
- Drag and Drop a sample from the **Samples** tree (or from your Library) on the left into a voice in the Voices and Zones window.
- Explore your computer for an audio file (Wave, AIFF, MP3, REX 2). Drag and drop the file into a voice in the Voices and Zones window. This imports the sample and adds it to the voice.

► **To Add a Voice and a Sample Simultaneously**

There are several ways to add a voice and a sample in the Voices and Zones window:

- Drag and Drop a sample from the **Samples** tree (or from your Library) on the left into the empty area just beneath the list of voices.
This creates a new voice containing the sample you selected. The new voice's original key will be the sample's original key (or C3 if this information is unavailable). The voice's key range will be C-2 to G8.
- Explore your computer for an audio file (Wave, AIFF, MP3, REX 2). Drag and drop the file into any area in the Voices and Zones window.
This creates a new voice containing the audio file you selected. The new voice's original key will be the audio file's original key (or C3 if this information is unavailable). The voice's key range will be C-2 to G8.
- Drag and Drop a sample from the **Samples** tree, from your Library, or from your desktop onto a note on the keyboard at the bottom of the screen.
This creates a new voice containing the sample you selected. The new voice's original key will be whichever note you dropped the sample onto. The voice's high and low keys will also be whichever note you dropped the sample onto.

Mix/Tune Page

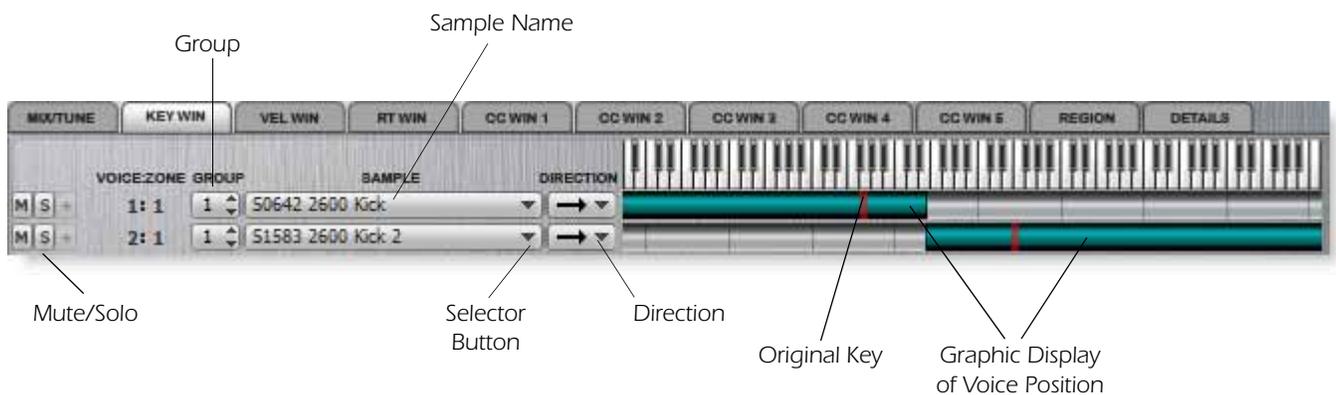


Volume, pan, direction, transpose, coarse tuning, and fine tuning can be adjusted separately for each voice. The Mix/Tune window allows you to see these settings for each voice at the same time, giving you an overview of your preset's direction, volume, pan, tuning, and transpose settings for each voice. All of these controls are available in the other tabbed windows, but in this window, you can see the volume and pan for each channel.

Additionally, from this screen, you can easily select multiple voices and change their volumes with one adjustment. Select the voices whose Volume or Pan you want to change, and hold **Shift** while dragging the volume right or left.

◆ See [page 129](#) to understand the difference between transpose and coarse tune. You can also set the group number from this page.

Key Window Page



When a sample is recorded, it is automatically placed in a voice and assigned a keyboard range specified in the "Auto-Placement" parameters in the "Acquire" or "Import Samples" options in the File menu. The Key Range Window allows you to edit these assignments and place the voices and samples wherever you want.

A voice containing only one sample takes the name of the sample. If a voice contains more than one sample, it is called a "Multisample" and the samples it contains are listed below it. The screen above shows two single sample voices. Voice 1 plays on the lower half of the keyboard and voice 2 plays on the upper half.

The **Original Key** determines the pitch of the sample. The original key need not be located between the high and low key ranges. Samples normally have the original key

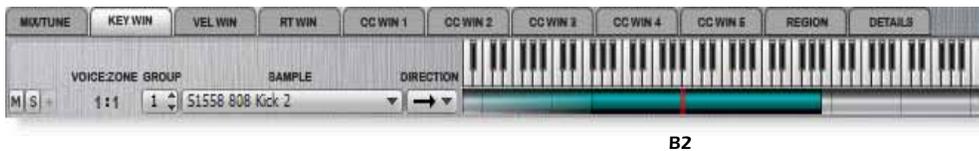
as part of the sample name. Imported samples with embedded original key information can be auto-mapped to the correct location. See [page 295](#).

The key position is edited by placing the cursor over the desired parameter and then playing the MIDI keyboard or adjusting the value using the keyboard. You can also simply drag the blue bar to the desired position with the mouse.

Voices and samples can be crossfaded according to key position by adjusting the low and high fade parameters. The fade is adjusted in “number of keys” and is graphically represented by a gradual decrease in the color intensity of the bars.

► **To Manipulate the Bars:**

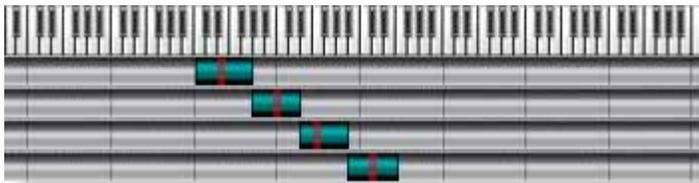
- To Set the Length of the Bar: Left-click and drag from either endpoint.
- To Move the Entire Bar: Left-click and drag from somewhere in the middle of the bar.
- To Create a Fade: Left-click and drag inward while holding the Alt key.



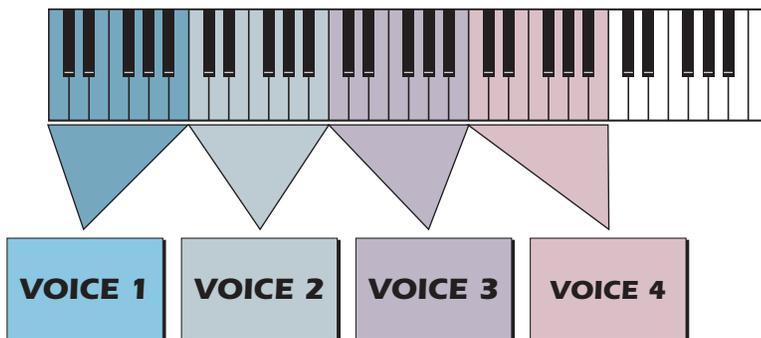
The voice will gradually fade-out in volume below B2 as you play down the keyboard.

► **To Switch Voices According to Position**

Assigning voices (samples or multisamples) adjacent to each other is the simplest and most common application of the Key Window. The voices in the screen below are each assigned a one octave range.



Another way of visualizing this assignment is shown below.



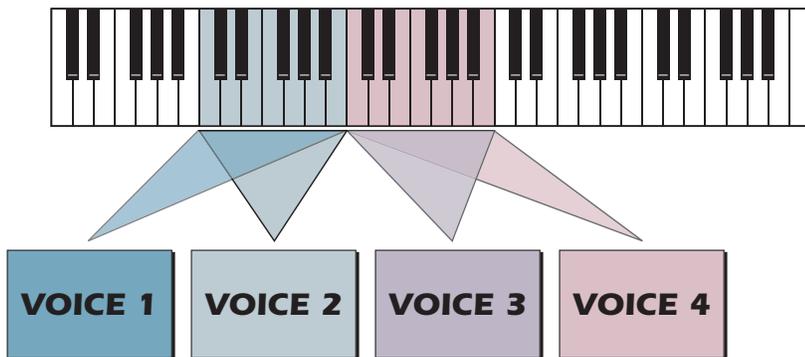
■ Samples can be arranged in a multisample voice in exactly the same way that voices are arranged on the keyboard. They can be arranged side by side on the keyboard or stacked on top of each other to create layered sounds, all within a single voice.

► **To Layer Voices and Samples**

The ranges of two or more voices may overlap, and all voices assigned to a key will sound.



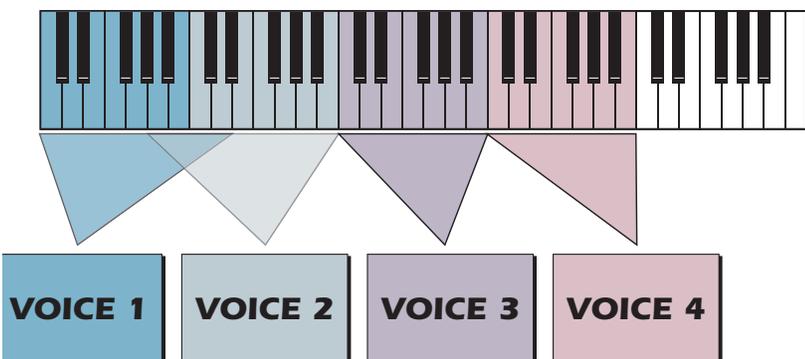
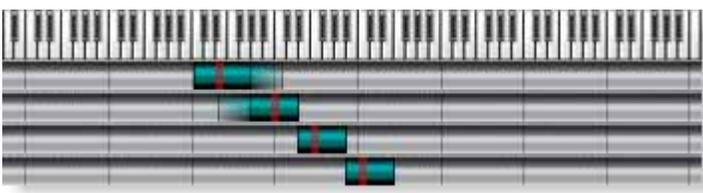
Another way of visualizing this assignment is shown below.



► **To Positionally Crossfade**

This technique can help mask the point at which two sounds meet or can be used for crossfading sound effects. The fade parameters allow the volume of the voice or sample to be faded out as a function of key position. In the screen shown below, voices 1 and 2 are being positionally crossfaded over a range of five notes.

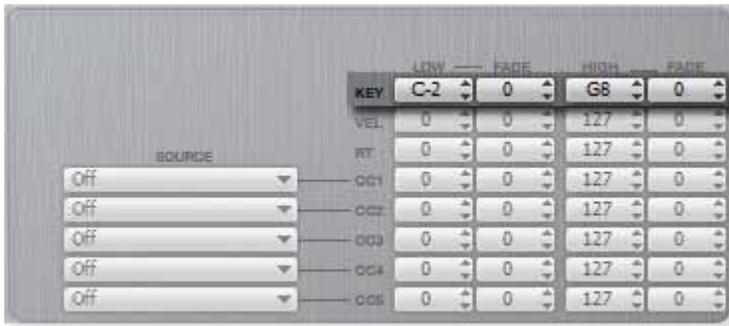
As the voice boundaries are crossed playing up the keyboard, the lower voice gradually fades out as the upper voice fades in.



◆ You may want to assign voices in different layers to different groups. This allows you to select all of the voices in each layer just by selecting the group number. You can edit the group number of a voice in any of the Voice screens.

Using Values to Define Key Window Parameters

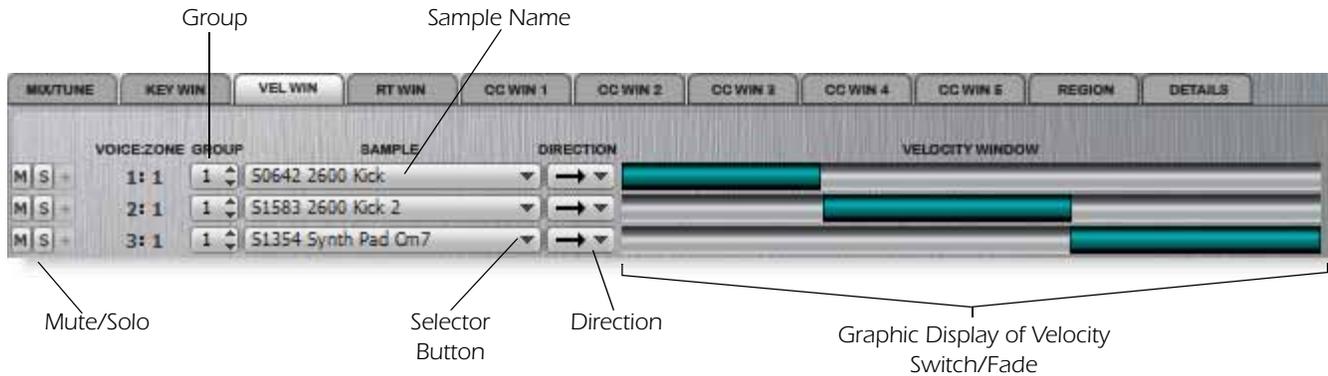
As you use the graphical interface to change the Key Window parameters, you'll notice that the values in the Overview section at the bottom of the screen change as well.



You can adjust the Key Window information here. Choices include High and Low notes, and High and Low fades.

Velocity Window Page

The Velocity Window allows you to control the volume of a voice by the key velocity. Using this function, you can crossfade or cross-switch between voices by key velocity. Suppose you had several samples (voices) of the same piano note, one played soft, the next played medium hard, and the third played hard. Using this function, you could set the velocity at which each of the voices would sound, thus recreating the response of the actual instrument.



► To Velocity Switch Voices:

1. In order to velocity switch voices, they must first be assigned to the same key range. In the Key Window, set the high and low key range of the voices so that they completely overlap.
2. Select the Velocity Window button shown below. By default both voices will be assigned to the full 0-127 velocity range.
3. Set the velocity range so that each voice has its own velocity range.
(In the above example, a key velocity of 1-35 will play voice 1, a key velocity of 36-80 will play voice 2, and a key velocity of 81 or above will play voice 3.)

► To Velocity Crossfade Voices:

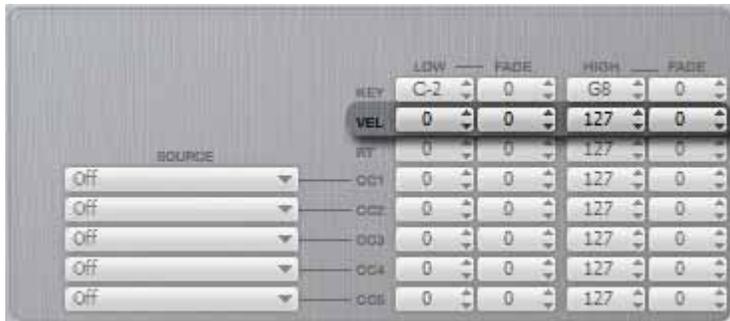
1. In order to velocity crossfade voices, they must first be assigned to the same key range. In the Key Window, set the high and low key range of the voices so that they completely overlap.



2. Select the Velocity Window button shown above. By default both voices will be assigned to the full 0-127 velocity range.
3. Set the Fade parameter for each voice so that as one voice fades out the other fades in.
4. Play the keyboard and fine tune the crossfade points for the desired response.

Using Values to Define Velocity Window Parameters

As you use the graphical interface to change the Velocity Window parameters, you'll notice that the values in the Overview section at the bottom of the screen change as well.

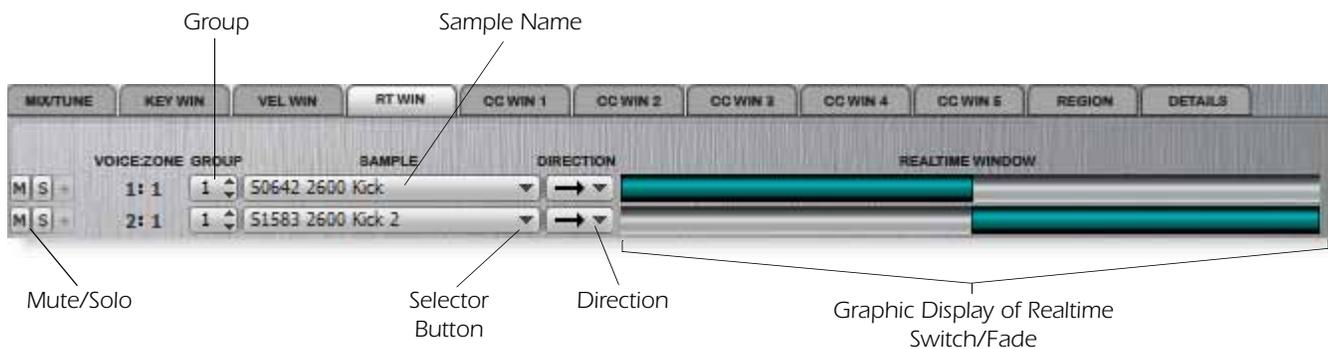


You can adjust the Velocity Window information here as well. Choices include High and Low notes, and High and Low fades.

Realtime Window Page

The Realtime Window allows you to control the volume of a voice using a real-time controller such as a modulation wheel, a pedal, an LFO or an envelope generator. It works in a manner very similar to the velocity window except that a realtime controller is used to control the volume instead of key velocity.

Note: Realtime cross-switching/crossfading of voices is inherently wasteful of polyphony because voice channels must be assigned even if the volume of a voice is at 0. (Since this is a *Realtime* control the volume can change at any time.) If you don't need realtime volume changes, you might want to investigate the ["Keyboard CC Window" on page 109](#).



► To Realtime Crossfade Two Voices:

1. In order to realtime crossfade voices, they must first be assigned to the same key range. In the Key Window, set the high and low key range of the voices so that they completely overlap.

▼ **Warning:** Make sure you understand how to select Voices and assign Cords before attempting to perform realtime crossfades.



Set up the Voices

2. Select the Realtime Window button shown above. By default both voices will be assigned to the full 0-127 range.
3. Set the Fade parameter for each voice so that as one voice fades out the other fades in. Voice 1 will be at full volume until the realtime controller reaches a value of 16. It will then begin fading out. Voice 2 will begin fading in immediately and be at full volume from 100 to 127.

Assign the Realtime Controller

4. Select one of the voices you want to crossfade by clicking anywhere on the strip. A red box forms around the voice to show that it's selected.
5. Now hold the **Ctrl** key and select the next voice you wish to crossfade by clicking on it. Both voices should be surrounded by red boxes.
6. With both voices selected, select **Open Voices** (**Ctrl+Enter**) from the Preset menu. The Voice Processing window appears. Verify that the two voices are selected in the "Voice Selector" section of the window.

7. Assign a realtime controller to control the crossfade. Select **Amp Crossfade** as the destination. Select a modulation source that goes from 0-127 such as Aux Envelope+, LFO+ or MIDI A-P. Set the **Cord Amount** to +127.
8. You should now be hearing the realtime crossfade when you play a key. You may have to go back to the Realtime Crossfade page to adjust and fine tune the crossfade points. These settings will vary with the sounds that are being crossfaded.

► **To Randomly Cross-switch or Crossfade Between Voices:**

In certain situations, such as setting up drum kits, you may want to randomly switch between several voices. Crossfade Random is a modulation source specifically designed to handle this situation. Unlike the other random sources, Crossfade Random generates one random number for all voices that are assigned to the same key.

1. In order to realtime crossfade or cross-switch voices, they must first be assigned to the same key range. In the Key Window, set the high and low key range of the voices so that they completely overlap.

Set up the Voices

2. Select the Realtime Window button.
3. Set up a basic cross-switch in the Realtime Window as shown below. For this example, we'll use two voices. This will cause voice 1 to sound whenever the random value is in the range of 0-63 and voice 2 to sound whenever the random value is in the range of 64-127.



Assign the Realtime Controller

4. Select the first voice by clicking anywhere on the strip. A red box forms around the voice to show that it's selected.
5. Hold the **Ctrl** key and select the next voice by clicking on it. Both voices should be surrounded by red boxes.
6. With both voices selected, select **Open Voices** (Ctrl+Enter) from the Preset menu. The Voice Processing window appears. Verify that the two voices are selected in the "Voice Selector" section of the window.
7. Assign **Crossfade Random** as modulation source in the Cords section. Select **Amp Crossfade** as the destination. Set the **Cord Amount** to +127.
8. Voices 1 and 2 will now be randomly selected whenever you press a key.

■ **Tip:** Random Cross-Switch can be used in combination with Velocity Switching to play random voices when you play softly and a known voice when you play hard.

Using Values to Define Realtime Window Parameters

As you use the graphical interface to change the Realtime Window parameters, you'll notice that the values in the Overview section at the bottom of the screen change as well.

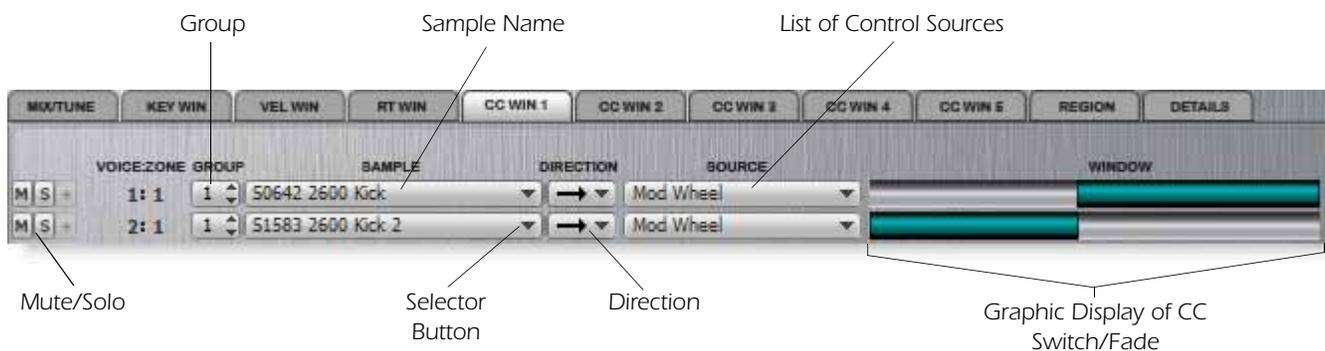


You can adjust the Realtime Window information here as well. Choices include High and Low notes, and High and Low fades.

Continuous Controller Window Pages

The CC Windows (1-5) offer additional voice switching and crossfading possibilities using MIDI Continuous Controller messages. Unlike the Realtime window, the CC crossfades and cross-switches occur only at Note-on time. They cannot be continuously swept during the note. These controls are designed to be used to switch and fade voices during your performance. You can use the CC windows to switch between drum sets, or bring in accent samples.

The CC assignments are easy to use because you can complete the entire patch from this window. The list of control sources is accessed by clicking on the button  under the Source column. The following control sources are available: Mod Wheel, Pressure, Pedal, MIDI A-P, Footswitch 1 & 2, Flip-Flop Footswitch 1 & 2, Thumbswitch and Flip-Flop Thumbswitch, Keyboard, Cycle 2-3-4 (Chan), Cycle 2-3-4 (Key), Random.



► To Cross-switch Two Voices:

1. In order to cross-switch voices, they must first be assigned to the same key range. In the Key Window, set the high and low key range of the voices so that they completely overlap.
2. Set up a basic cross-switch in one of the CC Windows as shown above. For this example, we'll use two voices. This will cause voice 1 to sound whenever the CC value is in the range of 0-63 and voice 2 to sound whenever the CC value is in the range of 64-127.
3. Select one of the controllers from the list. Do this for each voice you wish to switch. In the example above, voice 1 will sound whenever the Mod Wheel is less than halfway up. Voice 2 will sound whenever the Mod Wheel is more than halfway up.

Keyboard CC Window

The keyboard CC window is a great performance feature, which allows you to switch CC windows using specified keys on your MIDI controller. For example, you can have three different voices assigned to the keyboard, such as piano, organ and a horn section, and select each of the three sounds using the lowest three keys on the controller.

When you select the Keyboard option in the “CC Win” list, a section of the keyboard can be used to conditionally decide whether a voice is played (similar to the Key Win, Vel Win, and other CC Wins).

The second part of the feature appears on the Preset Globals page. Pressing the “Key CC Win” button, and setting a Keyboard Range field marks off a section of the keyboard to be used as control keys for the CC Window. The keys in this range will not sound.

Using the example in the first paragraph, you might mark off the three lowest keys on the controller as control keys, so they don’t sound notes when pressed. They are now used for selecting the different voices.

Once you specify a Keyboard CC Value range, those values will not trigger note events on that channel. No voices (no sound) will be played when you play a key in the specified range. Once you press the key in the specified range, that key becomes the current Keyboard CC Win value. You don’t have to hold the note. A new Keyboard CC Win value is set when you press another key in the specified range.

The Keyboard CC Win Key defaults to the lowest control key when the preset is selected. You don’t have to press a control key to hear sound.

When using the File Converter to convert Gigasampler files to EXB files, the “Keyboard Dimension” is converted to a Keyboard CC Win. The “Dimension Key Start” and “Dimension Key End” parameters in Gigasampler are translated to “Keyboard CC Win Key Start” and “Keyboard CC Win Key End”. These values appear in the dialog managing these values. Each Keyboard dimension split is mapped to one key when converting to a Keyboard CC Win.

► To Program a Preset using the Keyboard CC Window:

For this example we’ll use preset 000:010 **And Voice** from the Proteus X Composer bank. This preset has two different sounds, piano and voices, layered over the same keyboard range. You could also use any other preset that has layered voices.

1. Load the **Proteus X Composer** (or **Proteus Composer**) bank.

Reserve the Control Keys.

2. From the Tree select preset **000:010 And Voice**. The Preset Globals page appears.
3. Click the **Key CC Win** button (located in the upper left corner of the screen).
4. Set the **Low and High range keys** to the lowest two white keys on your keyboard. (Hint: Select the key field. Press and hold **Ctrl + Alt**, then play the keyboard to set the key.)

Set the CC Window.

5. Open the **Voices and Zones** page of **And Voice**.
6. Click the **CC Win 1** button.
7. For each voice, click the **Assign** button and select **Keyboard**.
8. Set the **High and Low keys on voice one** to the **lowest key** on your keyboard. (Don’t change the fade fields.)
9. Set the **High and Low keys on voice two** to the **next-to-lowest white key**.

10. **Play the keyboard.** Pressing the lowest key on the keyboard selects the piano. Pressing the next white key selects voices.
11. **Bonus Step:** You can also make the black key between the white trigger keys play both voices. Extend the High CC range of voice one up 1/2 step. Extend the Low CC range of voice two up 1/2 step. Playing the sharp now plays both voices.

Cycle Groups

The Cycle Group destinations apply rotational (or random) input controllers to the CC Window. Because the generated values are sources for the CC Windows, crossfades and complex cross-switching are possible.

This feature is very exciting because it allows for more realistic and natural-sounding presets than previously possible with a sampler. When you hear, for example, the same snare drum sample throughout a piece of music, it's very obvious that you're listening to a sampler, not a real drummer. But if you alternate between three or four similar, but not identical snare samples, the increase in realism is quite dramatic. Brass and string samples also benefit greatly from Cycle Groups.

In the simplest case, you can use this feature to alternate between two different voices each time you press a key (Cycle 2-Key). The first time a you press a key, a value of 0 is generated. The next time you press the same key, a value of 127 is generated, then 0-127-0-127, etc. These values are used as control sources for the CC Window to determine which voice will sound. Cycle 3 and Cycle 4 groups sequence three and four values respectively.

Channel Cycle Groups Rotate the value each time a note is played on the MIDI **Channel**.

Key Cycle Groups Rotate the value each time a note is played on the same **Key**.

- **Random** At note on, per channel, a random value between 0 & 127 is generated as the source for the CC Window.
- **Cycle 2 (Channel)** A value of **0 or 127** is generated each time a note event occurs on the **Channel**. The two numbers alternate each time a note event occurs on the channel. This value is used as the source for the CC Window.
- **Cycle 3 (Channel)** A value of **0, 64 or 127** is generated each time a note event occurs on the **Channel**. The generated value cycles sequentially through 0, 64 and 127 with each new note event on the channel. This value is used as the source for the CC Window.
- **Cycle 4 (Channel)** A value of **0, 42, 85 or 127** is generated each time a note event occurs on the **Channel**. The generated value cycles sequentially through 0, 42, 85 and 127 with each new note event on the channel. This value is used as the source for the CC Window.
- **Cycle 2 (Key)** A value of **0 or 127** is generated each time a note event occurs on the **Key**. The two numbers alternate each time a note event occurs on the key. This value is used as the source for the CC Window.
- **Cycle 3 (Key)** A value of **0, 64 or 127** is generated each time a note event occurs on the **Key**. The generated value cycles sequentially through 0, 64 and 127 with each new note event on the key. This value is used as the source for the CC Window.
- **Cycle 4 (Key)** A value of **0, 42, 85 or 127** is generated each time a note event occurs on the **Key**. The generated value cycles sequentially through 0, 42, 85 and 127 with each new note event on the key. This value is used as the source for the CC Window.

► Setting up the Cycle Groups

Here's a simple example to show you how to set up the cycle groups. In preparation, load the **Proteus X Composer** bank, open the **Presets** folder, and select preset **000:035 Fing&Harms** from the tree.

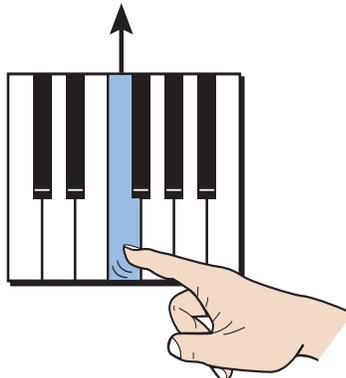
1. Expand the preset in the tree by clicking the + symbol next to the keyboard icon.
2. Click **Voices and Zones**. The Key Window appears with two voices (HardFinger A#0 and BassHarm G4).
3. For each voice, extend the Key Range so it covers the entire keyboard. (drag the bars)
4. Next, select new samples for each voice. (Click on the sample button and choose a new sample). It doesn't matter which two samples you choose as long as they are clearly distinguishable from each other.
5. Click the **CC Win1** button. The Continuous Controller Window 1 appears.
6. For each voice, click the **Assign** buttons and select **Cycle 2(Key)**.
7. Set the CC Window range for voice 1 by dragging the right end of Voice 1's bar to the left. It doesn't matter how far you drag it, as long as it doesn't include the value of 127.
8. Set the CC Window range for voice 2 by dragging the left end of Voice 2's bar to the right. It doesn't matter how far you drag it, as long as it doesn't include the value of 0.
9. Play the keyboard. The voices alternate each time you play the same key.
10. For each voice, select **Cycle 2(Channel)**. Now the voices alternate whenever you play a new key on the channel, regardless of which key you are playing.

Cycle 2 (Key)



CC Win Value = 0 - 127 - 0 - 127...

Cycle 2 (Key):
Generates alternating values of 0 & 127 on each key press



The Cycle 2 (Key) Group allows you to alternate between 2 voices with each key press.

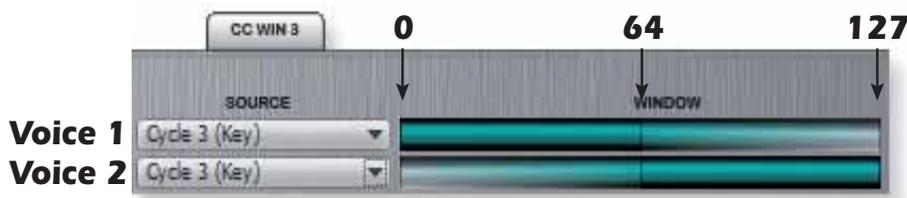
Cycle 3 Crossfading

This next example demonstrates the Cycle 3 mode as well as the crossfading abilities of the CC Window. Pressing a key will first play **Voice 1**. Pressing the key again will play a **50-50 mix** of both voices. Pressing the key a third time will play **Voice 2**. The next key press will play Voice 1 again and the cycle repeats.

■ **Tip:** Conserve polyphony by cross-switching, rather than crossfading, voices.

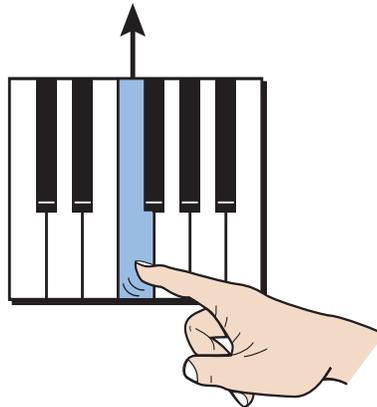
11. Set both **Voice 1** and **Voice 2** to **Cycle 3 (Key)**.
12. Create a fade in **Voice 1** like the one shown in the diagram below. **Alt-click** on the right end of the bar and then dragging to the right, then all the way to the left.
13. Create a fade in **Voice 2** so that it looks like the diagram below.
14. Repeatedly press a single key and listen to the voices rotate. The rotation order is retained for each key.

Cycle 3 (Key)



↑
CC Win Value = **0 - 64 - 127 - 0 ...**

Cycle 3 (Key):
Generates rotating values of 0 - 64 - 127 on each key press



- 1st key press = **Voice 1**
- 2nd key press = **Voice 1 & 2**
(50-50 Mix)
- 3rd key press = **Voice 2**
- 4th key press = **Voice 1**

⋮

Cycle 3 (Channel)

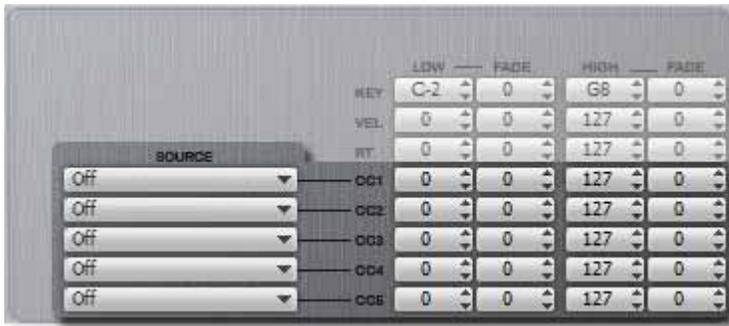
15. Assign both voices to **Cycle 3 (Channel)**.
16. Play the keyboard. Notice that the voices now rotate regardless of which key is played. Any key on the current MIDI channel will increment the rotation.

Random Mix

17. Assign both voices to **Random**.
18. Play the keyboard. Now the mix of the two voices is completely random.
19. You can "weight" the randomness by adjusting the crossfade points. For example, if you want Voice 1 to be dominant, adjust the fade of the voice so that most of the bar is solid. This way it has more chance of being played at full volume.

Using Values to Define CC Window Parameters

As you use the graphical interface to change the CC Window parameters, you'll notice that the values in the Overview section at the bottom of the screen change as well.



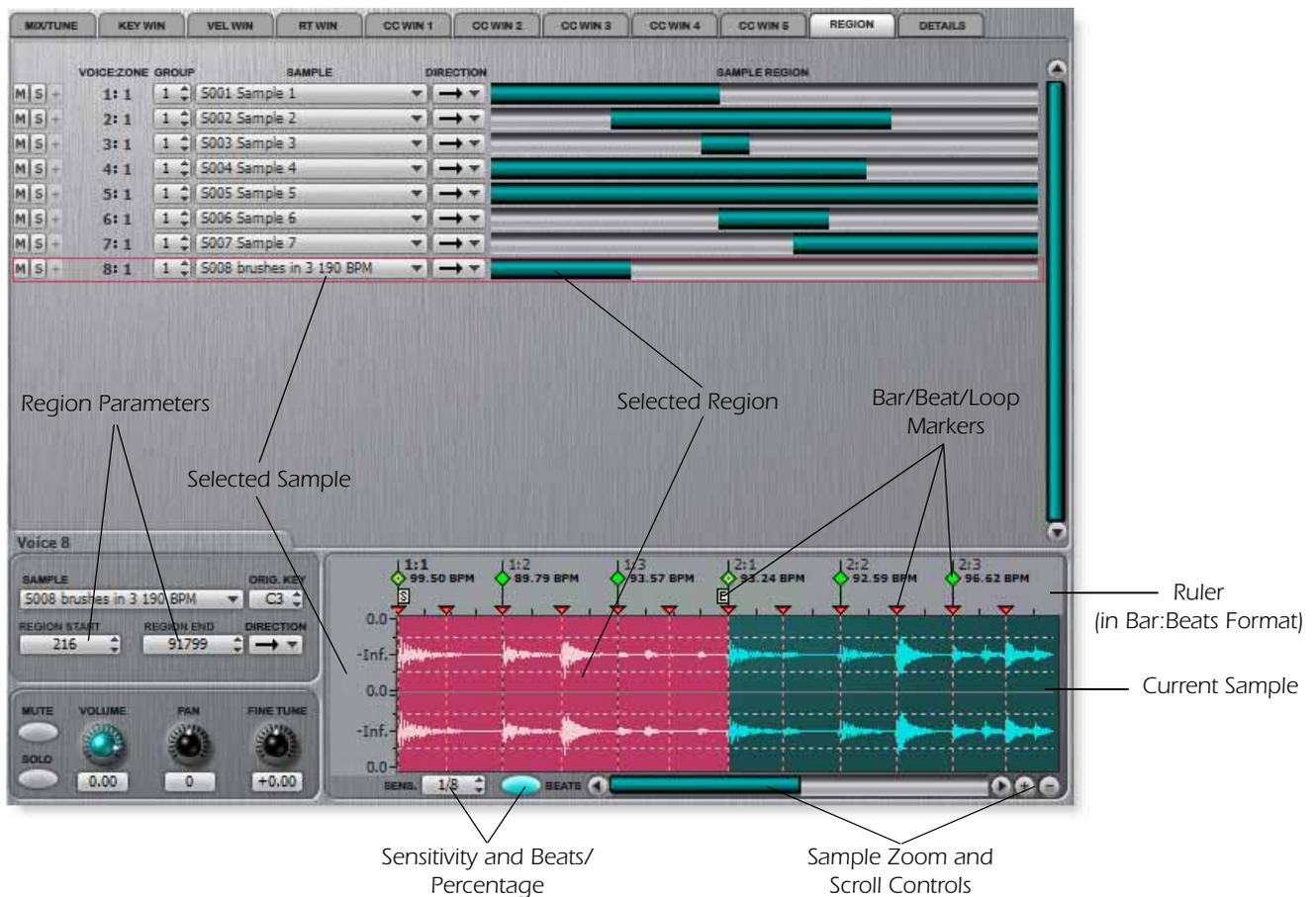
You can adjust the CC Window information here as well. Choices include Source, High and Low notes, and High and Low fades.

Region Page

The Region window allows you to define which part of a sample plays. In essence, it allows you to non-destructively crop the beginning and end of a sample.

- In its simplest use, you can re-define the start and end points of a sample.
- You can use the Region page to view sample information, such as bars, beats, and loops.
- You can define areas to loop forward or backward.
- You can Xplode a region, creating new voices from all visible note markers.

In the case of looping, you can view Beat, Note, and Loop markers, and snap to them. This way, you can loop portions of a sample in a similar way that you would with Twistaloop. The difference here is that if you have multiple voices using the same sample, you can set up each voice to play a different section (such as a Loop) of the same sample.

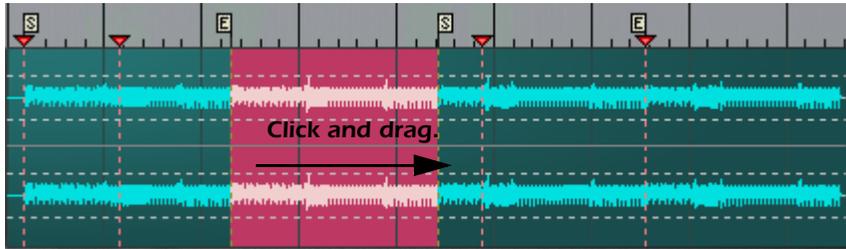


When you select a voice, or a sample in a multisample voice, the sample shows up in the bottom of the screen. By default the entire sample is defined as a region. The selected region is defined in pink.

Creating or Editing a Region

Creating or editing a region is very simple. There are three ways to do so:

- Click and drag in the display area.



■ **Tip:** To select the whole region, hold **Alt** while clicking the Sample Region range bar. This restores the default setting, which is ALL.

- Use the selection bars next to each voice or sample zone.



- Enter the information manually in sample values.



Region Start and End (in samples)

Region View Controls

Sample Zoom and Scroll Controls

You can use the Zoom In and Zoom Out Buttons to take a closer look at the sample. Use the Scroll Control to scroll backward and forward.



Sensitivity

The sensitivity setting determines the resolution of how Note Markers will be displayed. You can set the sensitivity in percentage from 0 to 100:



Or, if you click the **Beats** button, you can set the sensitivity ranging from eight-bar intervals (8/1) to sixteenth note intervals (1/16):



For example, if you set the sensitivity to 1/8, Note Markers fall on note onsets that appear near eighth notes. Similarly, setting the sensitivity to 1/16 causes Note Markers to find onsets near sixteenth notes.

- A letter “t” next to a sensitivity setting indicates triplets.
- A letter “d” next to a sensitivity setting indicates duplets.

Snap Settings and Markers

When you use your mouse to define a region, the region start and end points snap to either samples, Loop Markers, or Beat and Note markers depending on your view settings.

If you do not view any markers, region selection does not snap. With markers present, region selection snaps to whatever markers are present.

View Beat Markers From the toolbar, click  to toggle Beat Markers. With this view enabled, region selection snaps to beat markers.

View Note Markers From the toolbar, click  to view Note Markers. With this view enabled, region selection snaps to Note Markers.

View Loop Markers From the toolbar, click  to toggle Loop Markers. With this view enabled, region selection snaps to Loop Markers.

You can also select a region using the Sample Region Bars to the right of each sample. This method of selection does not snap. Click and drag the right or left edge of the bar to change the start or end of the region. Click and drag the center of the bar to move the entire region forward or backward.

Region Xplode

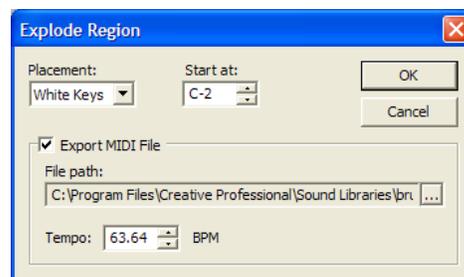
This option creates new voices from the area between each visible note marker in the selected Region. The new voices are mapped either chromatically or on sequential white keys.

Optionally, you can create a MIDI file that maps the new regions to MIDI notes, which you can use in a sequencer to play back each region consecutively.

You can only Xplode a single-sample zone, and the region you Xplode must contain note markers.

► To Xplode a Region

1. Select the voice whose region you want to Xplode.
2. Select the sensitivity you want. This determines the length of each segment into which the region is divided. Remember, the region is divided depending on how many note markers are visible.
3. From the Preset menu (or when you Right-click the Region), select **Region Xplode**. The following dialog box appears:



4. Select how you want the new voices to be mapped from the **Placement** drop-down menu:
 - **White Keys** maps each voice sequentially to a white key, beginning at the key selected in the **Start at** field.
 - **Chromatic** maps each voice sequentially chromatically, beginning at the key selected in the **Start at** field.
5. Under **Start at**, enter the lowermost note where you want placement to start.
6. Decide whether you want to export a MIDI file. If so, check **Export MIDI File**, and choose a location and tempo (by default, the tempo is filled in as the tempo of the first note, which is usually the best choice).
7. Click **OK**.

You should now have new voices, each using the same sample, but different regions. It might look something like this:



More About Regions

You should remember that if a sample contains loops, selecting a region does not override these loops; Selecting a region merely determines which portion of a sample is allowed to play.

However, if you select a region that excludes part of a loop, such as the start or end point, that loop will not repeat.

Warning: If a region contains only a part of the sample, DO NOT edit the sample in the Sample Editor (e.g. Time Stretch, Sample Rate Convert, etc.). Doing so will cause region information to become incorrect. This does not apply if a region contains the *entire* sample.

Since regions contain the entire sample by default, this is only a potential issue if you have created a region. To ensure a region contains an entire sample, you can always **Alt-Click** the sample region bar to restore the region to its default setting.

Details Page

The screenshot shows the 'DETAILS' page of a software interface. At the top, there are tabs for 'MIX/TUNE', 'KEY WIN', 'VEL WIN', 'RT WIN', 'CC WIN 1', 'CC WIN 2', 'CC WIN 3', 'CC WIN 4', 'CC WIN 5', 'REGION', and 'DETAILS'. The 'DETAILS' tab is active, displaying a table with the following columns: 'M S +', 'VOICE:ZONE', 'GROUP', 'SAMPLE', 'DIRECTION', 'KEY WIN' (subdivided into 'LOW', 'FADE', 'HIGH', 'FADE'), 'OR KEY', 'VOL', 'PAN', 'X'POSE', 'COARSE', and 'FINE'. The table contains 8 rows of data, with the 8th row highlighted in red. Labels with arrows point to various elements: 'Group' points to the 'GROUP' column; 'Sample Name' points to the 'SAMPLE' column; 'Original Key' points to the 'OR KEY' column; 'Mute/Solo' points to the 'M S +' column; 'Selector Button' points to the up/down arrows in the 'GROUP' column; 'Direction' points to the 'DIRECTION' column; 'Key Win Parameters' points to the 'LOW', 'FADE', 'HIGH', 'FADE' sub-columns; and 'Mix/Tune Parameters' points to the 'VOL', 'PAN', 'X'POSE', 'COARSE', and 'FINE' columns.

M S +	VOICE:ZONE	GROUP	SAMPLE	DIRECTION	KEY WIN	OR KEY	VOL	PAN	X'POSE	COARSE	FINE			
					LOW	FADE	HIGH	FADE						
M S +	1: 1	1	5001 Sample 1	→	C-2	0	G8	0	C3	0.0	+0	+0	+0	+0.00
M S +	2: 1	1	5002 Sample 2	→	C-2	0	G8	0	C3	0.0	+0	+0	+0	+0.00
M S +	3: 1	1	5003 Sample 3	→	C-2	0	G8	0	C3	0.0	+0	+0	+0	+0.00
M S +	4: 1	1	5004 Sample 4	→	C-2	0	G8	0	C3	0.0	+0	+0	+0	+0.00
M S +	5: 1	1	5005 Sample 5	→	C-2	0	G8	0	C3	0.0	+0	+0	+0	+0.00
M S +	6: 1	1	5006 Sample 6	→	C-2	0	G8	0	C3	0.0	+0	+0	+0	+0.00
M S +	7: 1	1	5007 Sample 7	→	C-2	0	G8	0	C3	0.0	+0	+0	+0	+0.00
M S +	8: 1	1	5008 brushes in 3 190 BPM	→	C-2	0	G8	0	C3	0.0	+0	+0	+0	+0.00

The Details page allows you to view, in rows, almost all of the Voices and Zones information for all voices at once. For each voice, from left to right, you can view and edit Group, Sample, Direction, Key Win, Original Key, Volume, Pan, Transpose, Coarse Tune, And Fine Tune.

► Tips on using the Details Page:

- While the Mix/Tune, Key Window, and Velocity Window pages give you graphic displays of settings such as volume and pan, you can be very precise with your settings in the Details page.
- You can make very precise relative changes from the Details page. Try selecting multiple voices and using the  buttons to change a value. All selected voices' values will change in a relative manner. You can get more granularity this way when making precise settings with multiple items selected.

Voice & Sample Zone Utilities

These utilities are invaluable when creating and modifying Emulator X voices. The voice and Sample Zone Utilities are located under the Preset menu.

Open Voices

This function opens the Voice Processing window for the currently selected voice(s).

Open Sample

This function opens the Sample Editor for the currently selected sample.

New Voice

This function allows you to add a new empty voice to the preset. You'll use this function when building new presets from scratch or to add a voice to an existing preset.

Delete Voices

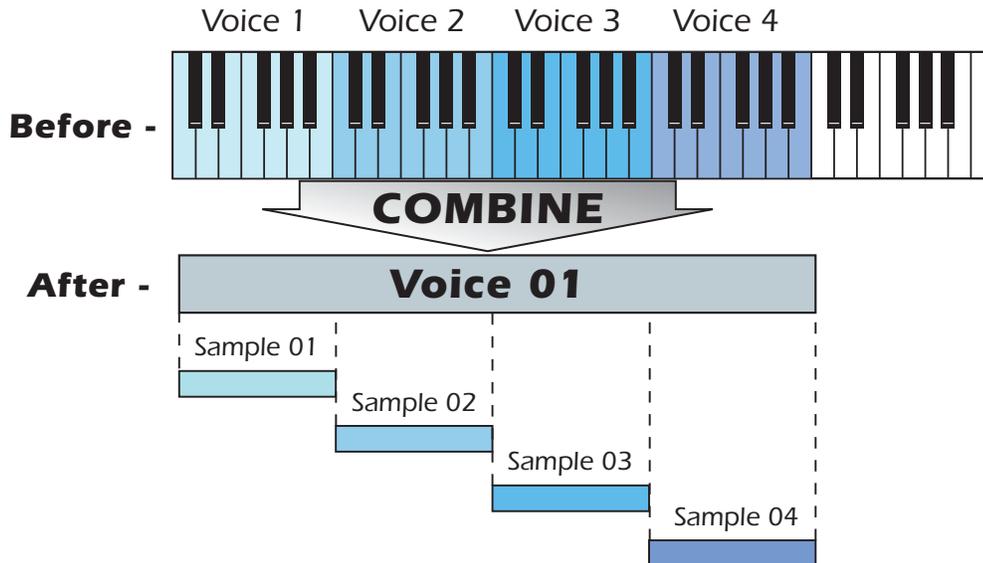
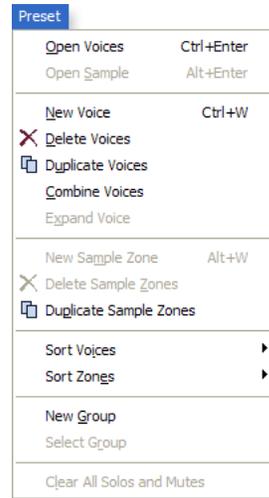
This function deletes the selected voice, but does not erase the samples.

Duplicate Voices

This function makes a copy of the voice, but does not copy the samples.

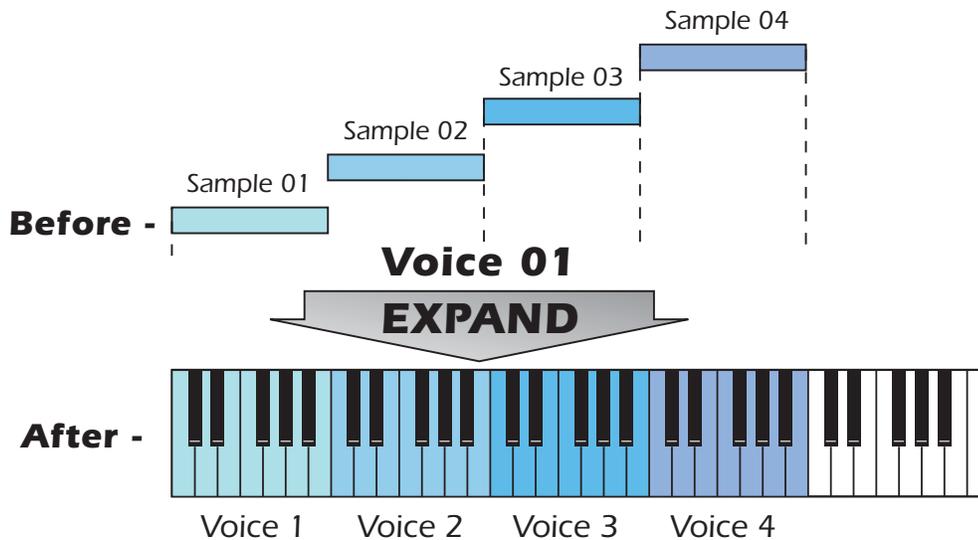
Combine Voices

This function takes all selected voices and combines them into one multisampled voice. The synthesizer level programming of the first voice is used for the new voice. Any synthesizer level programming associated with the other voices is lost.



Expand Voice

This function takes apart the selected multisampled voice and makes an individual voice for each sample. The voice processing information of the “parent” voice is duplicated for each newly created voice.



New Sample Zone

This function creates a new sample zone in the currently selected voice. If this operation is performed on a single sample voice, the voice is transformed into a multi-sample.

Delete Sample Zones

This function deletes the currently selected sample zones from the voice.

Duplicate Sample Zones

This function makes a copy of the currently selected sample zones, but does not copy the samples.

Sort Voices

You can sort voices by Sample Number, Sample Name, Low Key, High Key, Original Key, Volume, Pan, or Group.

Sort Zones

You can sort a zone by Sample Number, Sample Name, Low Key, High Key, Original Key, Volume, or Pan.

New Group

Adds all selected voices to a new group.

Select Group

Selects all voices that share the same group number with the currently selected voice. For example, if the voice you selected is a member of group 3, the **Select Group 3** command selects all other voices in group 3. This command is unavailable if you have multiple voices from multiple groups selected.

Clear All Solos and Mutes

Un-mutes and un-solos all voices and zones.

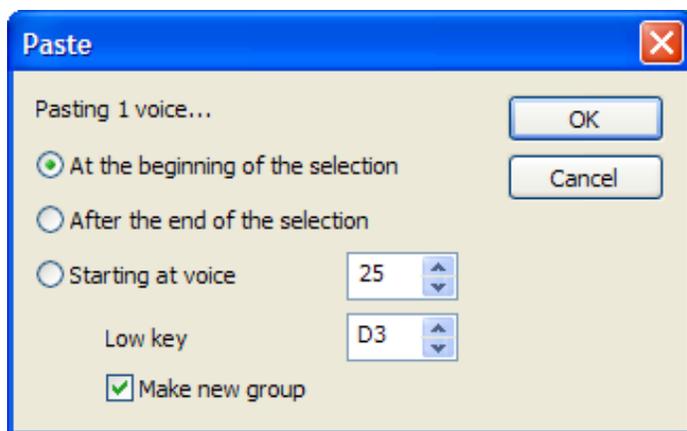
Edit Menu Utilities - Cut, Copy & Paste

You can use the standard cut, copy & paste utilities when working with voices and sample zones. Single or multiple voices can be selected, then moved or rearranged within the bank or between banks. This makes it easy to create or modify custom presets and banks.

◆ Voices cannot be pasted onto the desktop or outside of a bank.

► To Copy and Paste Voice(s)

1. Select the voice(s) you wish to copy. (Hold Shift to select contiguous voices.)
2. Select Copy from the Edit menu.
3. Open a new preset if you want to paste between voices.
4. Select the Voices and Samples page.
5. Select Paste from the Edit menu. The following dialog box appears.



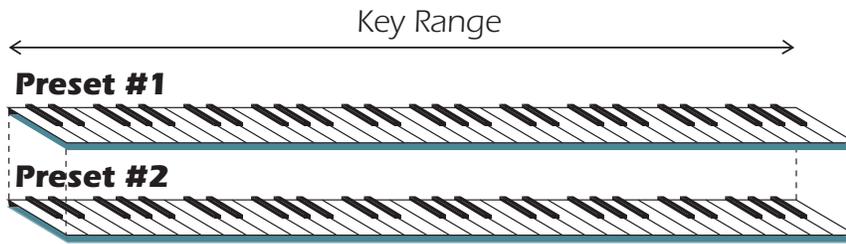
6. You have the option of pasting the voice(s) before the currently selected voice, after the currently selected voice or starting at a specified voice number. This feature allows you to rearrange your voices any way you want.
7. Press OK to paste the voice or Cancel to cancel the operation.

Links

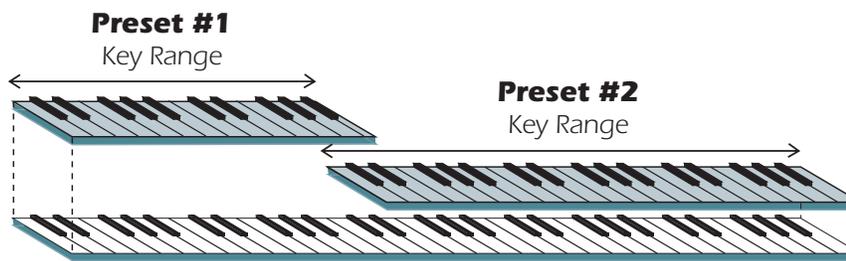
Links call up additional presets in the bank as a quick and easy way to create layering or keyboard splits. Linked presets are not altered by being linked to another preset, they are simply played.

One thing you need to be aware of when using links is that the Initial Controller settings (page 81) of the first preset are imposed on the linked preset. This may change the sound of the linked presets. Sometimes switching which preset is the Link will correct the problem. If this doesn't work you will have to edit the Initial Controller settings until both presets sound the way you want.

- You can quickly access the Links page for the current preset by clicking the  button in the toolbar.



Layering Two Presets Using Links

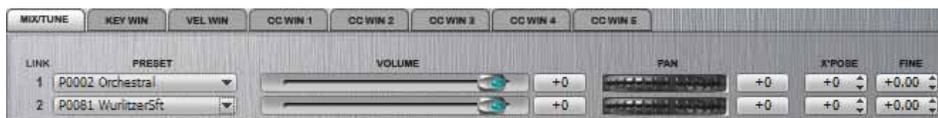


Creating a Split Keyboard with Links

One way to create new presets is to simply link several other presets to an empty preset. Each linked preset can be customized with its own keyboard range, velocity range, continuous controller assignments, volume, pan position, and tuning.

► To Access the Link Windows:

1. Select the **Sampler Tab** in the Tree.
2. Open the **Presets** folder by clicking on the (+) sign next to the folder.
3. Select a **Preset** or **Create a New Preset** (Presets > New Preset).
4. **Open the Preset** by clicking on the (+) sign next to the preset icon.
5. You'll see three sub-icons beneath the Preset icon. Click on the **Links** icon. The Links window opens.

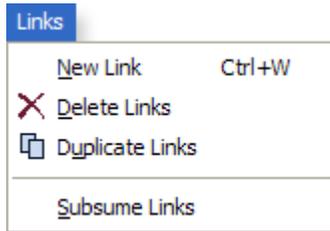


6. Select **Links > New** to add a new link.
7. The **Mix/Tune** window appears. Click the tabs across the top of the window to select the various Link windows.

Tip: You can also create a new Link by simply dragging and dropping a preset onto the preset field in the Link page.

Link Utilities

These four controls are accessed from the main menu.



- New Link** Creates a new link.
- Delete Links** Deletes the selected link(s).
- Duplicate Links** Duplicates the selected link(s).
- Subsume Links** Adds all voices from all links to the current preset and removes the link(s).

► To Create a Huge Stacked Preset the Easy Way:

1. Load any bank of presets.
2. From the **Sampler Tab** of the Tree, click on the **Presets** folder.
3. Select **Preset > New Preset** to create a new blank preset. "Blank Preset" will appear in the Tree. (Look at the bottom of the list.)
4. **Rename** your new preset and **double-click on the plus sign** to open it. The Voices & Zones, Links and Voice Processing branches appear.
5. Click on the **Links** icon. The Link window appears.
6. Select **Links > New Link** to create a new blank link.
7. Click on the **Preset button** to the right of the preset field. A pop-up menu appears.
8. **Select a preset.**
9. Repeat steps 6 and 7 to add additional presets until the desired thickness is achieved.

▼ If you link two or more presets, the Preset FX of the linked presets will be ignored. The linked presets will be routed through the effects of the main preset.

Subsuming Links

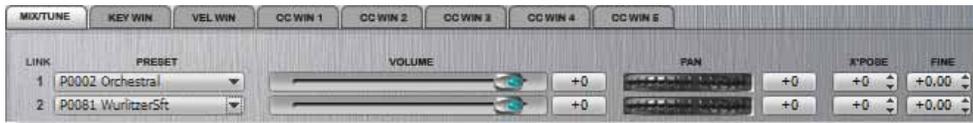
If you want to add all the voices from your links to the current preset, use the Subsume Links command. This adds all voices from the linked presets to the current preset, and removes the links.

For single-sample voices, subsuming links only brings in voices that are located in the range defined in the Links Key Window. If a voice falls entirely out of the range defined in the Links Key Window, it is discarded when you subsume links.

For multisample voices any sample whose voice falls within the range defined in the Links Key Window is brought into the preset, even if the sample itself is mapped to a location that falls out of the Links Key Window range.

Mix/Tune Window

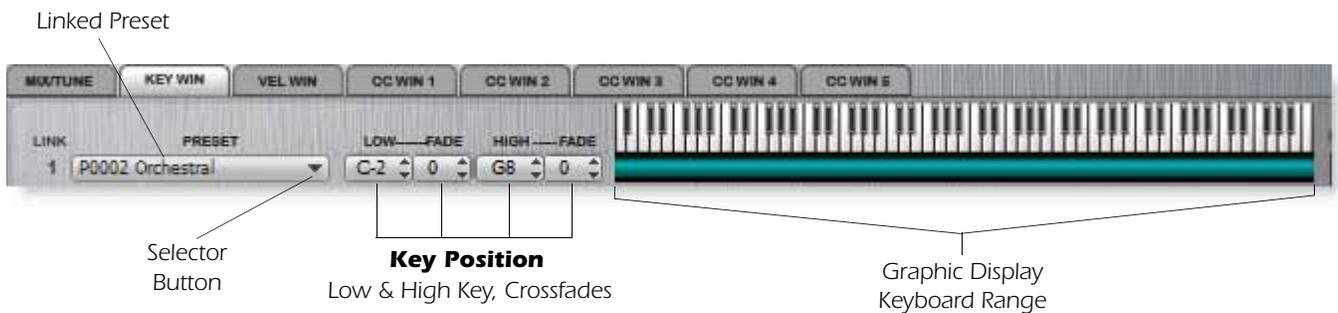
This window allows you to set the volume, pan position, transpose and fine tuning for the linked preset.



- Volume** Adjusts the volume of the linked preset.
- Pan** Sets the stereo pan position of the linked preset.
- Transpose** Transposes the linked preset by shifting the keyboard position. (See [page 129](#).)
- Fine Tune** Allows you to detune the linked preset by ± 100 cents. (1 semitone)

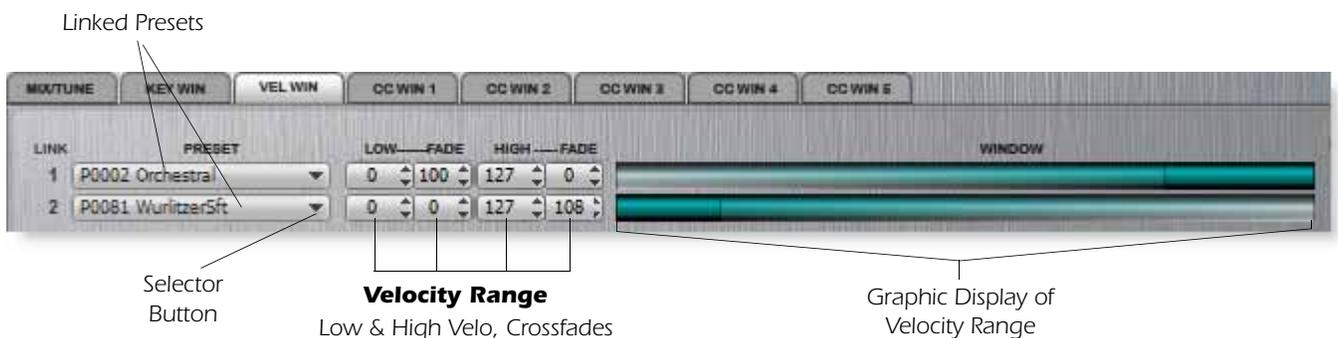
Key Window

The key window works just like the key window in the Voices and Zones section ([page 99](#)) and allows you to set the keyboard range for each linked preset.



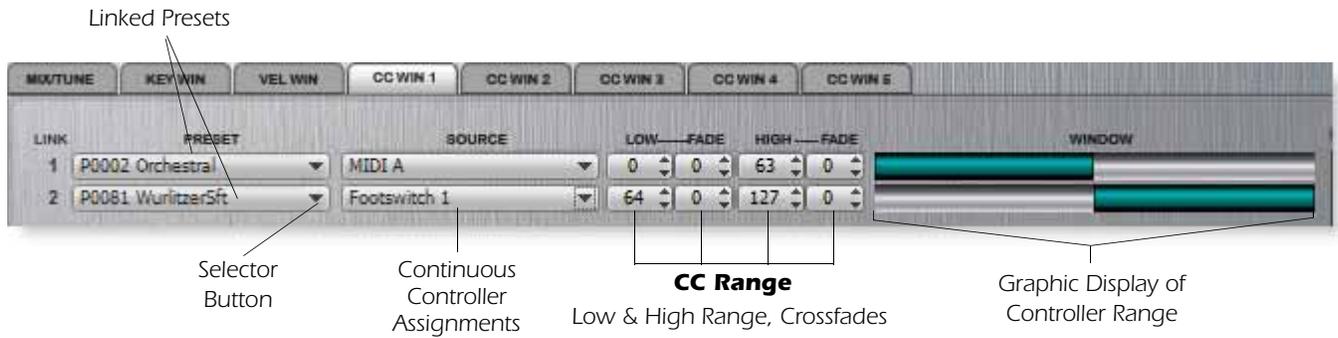
Velocity Window

The velocity window works just like the Velocity window in the Voices and Zones section ([page 103](#)) and allows you to set the velocity range for each linked preset. In the example below the presets are set to crossfade from Link 1 to 2 as the key velocity is increased.



CC Window

The CC or continuous controller window works just like the CC window in the Voices and Zones section ([page 108](#)) and allows you to set the range for each linked preset using continuous controllers. In the example below the presets are set to switch from Link 1 to 2 as the Mod Wheel is turned past 63. The CC values are locked in at note-on time.



6 - Voice Processing

Voice Processing contains the traditional synthesizer controls such as envelope shaping of amplitude and filters as well as modulation parameters. When multiple voices are selected, parameter changes affect all selected voices (i.e. setting the filter cutoff to 50, sets it to 50 on all selected voices). The voice processing parameters are accessed by clicking on the Voice Processing icon in the Tree.

► To Select Multiple Voices:

- Assign voices to Groups in the Voices and Zones page, then select the group.
- Select the desired voices from the Voices and Zones page.
- Hold Ctrl+Alt and play your MIDI keyboard to select voices (with IntelliEdit on).
- Press Ctrl-A to select all voices from the voice processing page.

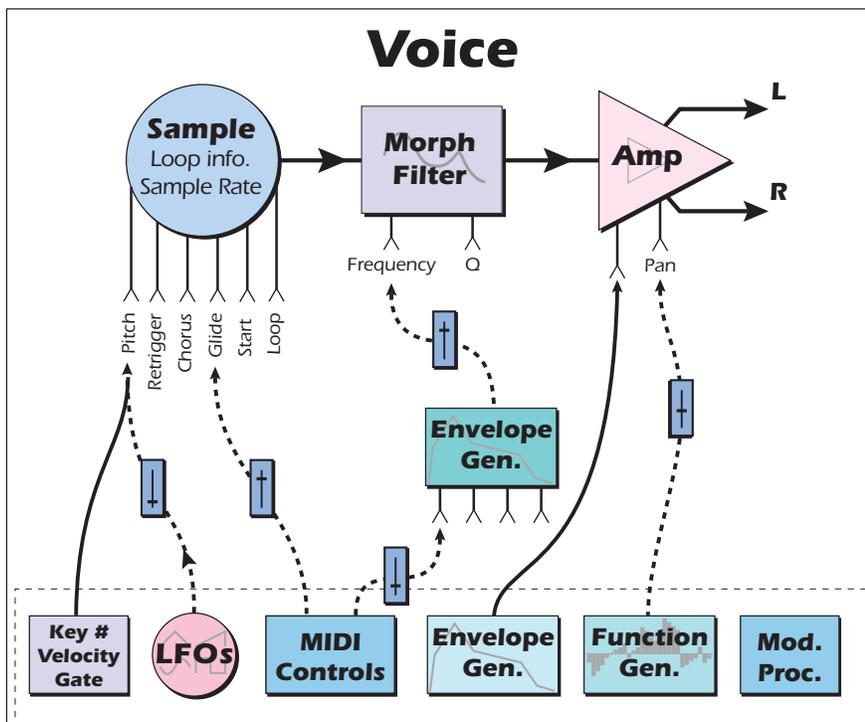
■ See Groups on [page 93](#).

■ IntelliEdit is enabled in the MIDI Preferences dialog box. See [page 26](#).

Simplified Emulator X Signal Path

A single Emulator X voice is diagrammed below. The audio path includes one or more samples, a dynamically controlled filter, and a dynamically controlled amplifier with a panning network. Modulation parameters operate simultaneously on all samples contained in a voice. PatchCords with built-in amount controls connect modulation sources (LFOs, envelopes, knobs, etc.) to modulation destinations (sample pitch, filter, amp volume, etc.). Memorize this voice architecture diagram to better understand Emulator X and make programming much easier.

■ You can quickly access the Voice Processing page for the current preset by clicking the  button in the toolbar.



Overview of the Voice Processing Window



The Voice Processing window displays all the synthesizer controls and PatchCord routings for the selected voice(s). Each module contains an independent synthesizer function.

Templates

For each different module in the Voice Processing window, you can save a library of your favorite settings. These Templates make it easy to create your own customized voices and presets. Templates are saved with the Emulator X application so they're always available for use. For more information on templates, see ["Templates" on page 279](#).

Oscillator



Pitch Bend Range

These two sliders allow you to set the amount of pitch bend up or down when a MIDI pitch wheel message is received. Pitch wheel messages are automatically routed to pitch unless these controls are set to zero.

TwistaLoop Controls

Speed

Please refer to [page 224](#) for detailed information about this control.

Loop (% or number)

Please refer to [page 223](#) for detailed information about the “Loop %” control.

Please refer to [page 217](#) for detailed information about the “Loop Number” control.

Start At Loop

Please refer to [page 217](#) for detailed information about the “Start At Loop” control.

Tuning Controls

Transpose

This function allows you to transpose the key of the selected voice(s) in semitone intervals by shifting the keyboard position relative to middle C. The transpose range is -36 to +36 semi tones. You will normally use the transpose function to tune a voice to its keyboard location, or simply to transpose the voice.

This is the same Transpose control displayed on the Mix/Tune tab of Voices & Zones ([page 99](#)) and changes made to either page will appear in both places.

Coarse Tuning

This function allows you to change the tuning of the voice in semitone intervals. The coarse tuning range is -72 to +24 semi tones. Unlike the transpose function, coarse tuning stretches the pitch of the individual samples assigned to a key and may change the timbre of a voice.

This is the same Coarse Tuning control displayed on the Mix/Tune tab of Voices & Zones ([page 96](#)) and changes made to either page will appear in both places.

- Use Transpose when you want the voice's timbre to remain constant.

- Use Coarse Tuning when you want to change the timbre of the voice.

- Use the Coarse Tuning control to change the tuning of voices containing multiple drums.

Fine Tuning

This function allows you to change the tuning of the voice in 1 cent intervals (100 cents = 1 semitone). The fine tuning range is ± 100 cents (± 1 semitone). Fine tuning can be used to slightly detune a voice, creating a “fatter” sound when it is combined with another voice.

This is the same Fine Tuning control displayed on the Mix/Tune tab of Voices & Zones (page 96) and changes made to either page will appear in both places.

Fixed Pitch Mode

This function turns keyboard transposition On or Off for the voice. With Fixed Pitch On, the keyboard will not control the pitch of the voice. This is a useful function for drones or “chiffs”, which you may not want to track the keyboard.

Chorus Amount



Chorus “thickens” the sound by doubling the sound in stereo and then detuning it. Chorus uses twice as many voices and twice as much of your CPU power when the Amount is set above zero.

Amount

The amount of detuning and is variable from Off to 100%.

Stereo Width

Controls the stereo spread. 0% reduces the chorus to mono and 100% gives the most stereo separation.

Inter-Aural Time Delay (ITD)

Adjusts the delay of the left and right sounds. Positive values start the right channel early and the left late.

▼ Chorus is automatically disabled on any voice that uses the TwistaLoop time modification feature (Lock to Tempo or TwistaLoop Speed). The Chorus controls can be set, but the effect will be disabled.

Chorus amount and ITD can be modulated in the Cords section, although ITD can only be changed at the time of the key depression. When Chorus is on, a mono voice will use twice as many channels.

“Off” is different than “0” for chorus amount. With mono samples, chorus On effectively turns a mono sample into a stereo sample with detuned L & R sides, with Chorus Amount being the detune amount. With chorus Off, no amount of modulation will turn the chorus on. With chorus On, modulation can drive the chorus (detune) amount to a minimum of 0, but not Off.

Tips for Using Chorus

- | |
|--|
| <ul style="list-style-type: none">▪ Set chorus amount to 1% and use chorus width as a pan control which does not span the entire L/R spread. |
| <ul style="list-style-type: none">▪ Set the chorus width to 0% and adjust the ITD to create a through-zero flange effect. With 100% width, a panning-chorus effect is obtained. |
| <ul style="list-style-type: none">▪ To avoid L/R pitch drift, use “~” type of modulation in the cords selection box. |
| <ul style="list-style-type: none">▪ Modulate the ITD to create “Hass-Effect” panning, (i.e. apparent location shift using slight delays in the L-R channels.) |
| <ul style="list-style-type: none">▪ For a detuned analog oscillator sound, set stereo width to 100%. Route a free running LFO (~) to chorus position with amount set to about 25%. |

Glide Rate & Curve

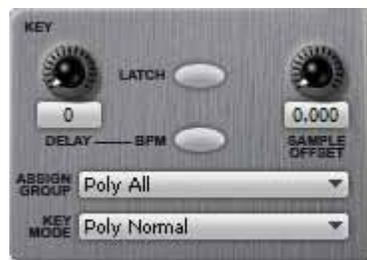


Glide is a smooth gliding between notes instead of the normal instantaneous change in pitch when a new key is pressed. The glide rate determines the time it takes to glide to the new pitch. The larger the value, the slower the glide rate. The rate is adjustable from 0 to 32.738 seconds. Zero seconds equals Off. The glide curve setting is adjustable between **linear** (large intervals result in longer glide time) and **logarithmic** (small or large intervals take roughly the same time). A logarithmic curve Glide is polyphonic, but can be monophonic when in Solo mode.

Key Controls

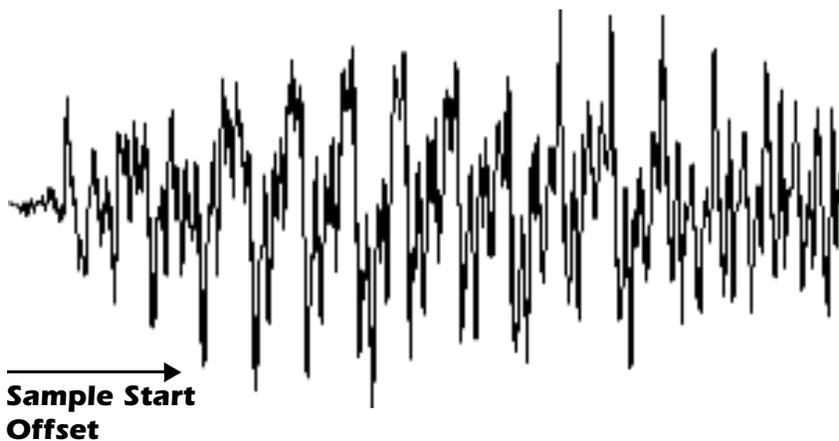
Delay

Delay varies the time between the arrival of a MIDI Note-On message and the onset of a note. The delay time is adjustable from 0 to 10,000 milliseconds (0-10 seconds). Delay can be used to create echo effects or to thicken the sound when using layered voices.



Sample Offset

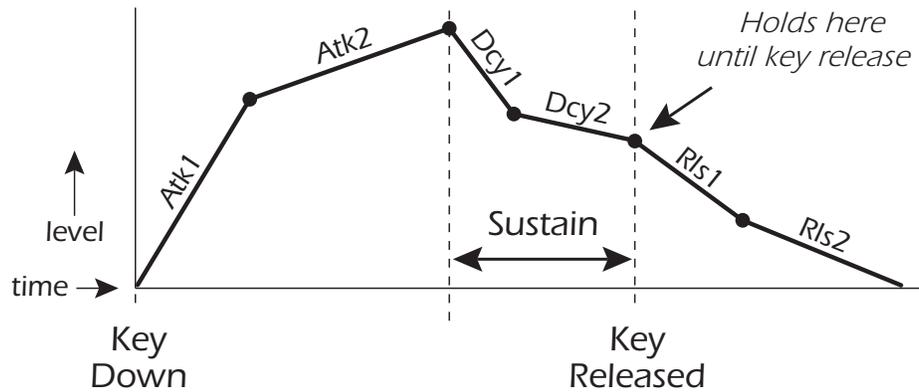
This function allows you to set where the voice's sample begins playing when you hit a key. A setting of 0 plays a sample from the beginning, higher values move the sample start point toward the end of the sound.



Important Note: If the Sample Start Offset is moved past the Pre-roll length, the sound will be cut off. If you are having Disk problems when using Sample Offset, increase the Pre-roll in the Preferences menu, or turn Streaming Off. [See "Pre-roll \(seconds\)" on page 22.](#)

Amplifier, Filter & Auxiliary Envelopes

Emulator X contains three envelope generators per voice. The Amplifier Envelope is “hard-wired” to control the Amp Volume and cannot go below the zero baseline (0-100). The Filter and Auxiliary Envelope generators are general purpose envelopes which must be patched to a destination using a PatchCord. The levels of the Filter and Auxiliary Envelopes can be programmed between the value range of -100 to +100.



Adjusting the Envelopes

There are four ways to adjust the envelope parameters:

- Adjust the Time/Level knobs.
- Grab one of the points on the Envelope Display and drag the envelope.
- Position the cursor in the Time or Level numeric fields and type in a time value.
- Position the cursor in the Time numeric field and type in the exact time.

The time values are calculated using time and the vertical distance to the next point.

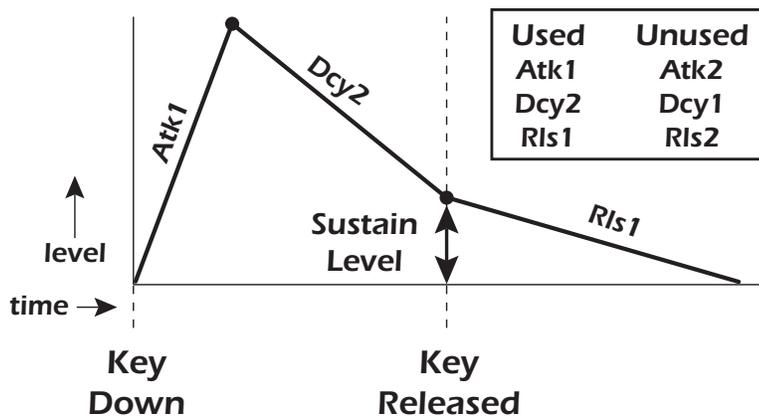


► **To Setup the Volume Envelope as an ADSR**

The classic synthesizer ADSR (Attack-Decay-Sustain-Release) envelope is easy to create using the six segment envelope generators. The trick is to only use segments: Attack1, Decay2, and Release1.

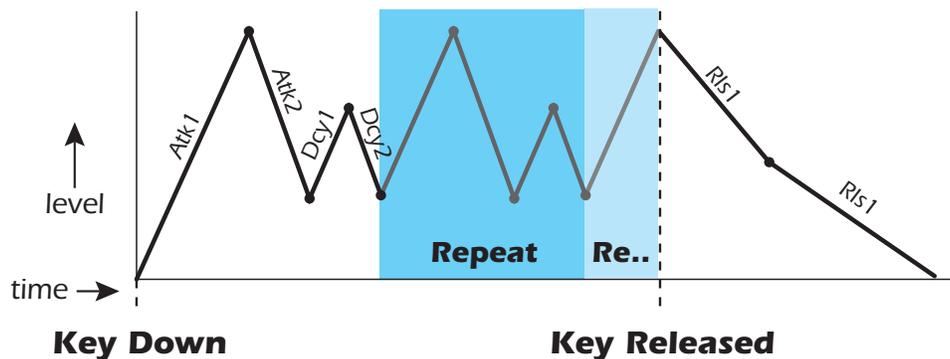
1. Set Atk2 level to 100.
2. Set all other levels to 0.
3. Set all times to 0.

Program the Atk1, Dcy2 and Rls1 time segments as you wish. Decay 2 level is the Sustain level. When programming controller knobs to the envelope times, you must use a negative cord amount in order to increase the time of an envelope segment.



Envelope Mode: Repeat

The filter and aux envelope generators can be made to repeat. When the envelope repeat function is On, the Attack (1&2) and Decay (1&2) stages will continue to repeat as long as the key is held. As soon as the key is released, the envelope continues through its normal Release stages (1 & 2).



The diagram above shows how the looping envelopes work. When the key is pressed the envelope goes through its regular Attack 1, Attack 2, Decay 1 and Decay 2 stages. In non-looping mode, the envelope would hold at the end of the Decay 2 stage until the key was released. In looping mode however, it jumps back to the Attack 1 stage and repeats the first four stages. It continues to loop through these four stages until the key is released where it immediately jumps to the Release 1 stage.

▼ The Amp Envelope does not have the repeating feature.

Envelope Mode: Complete

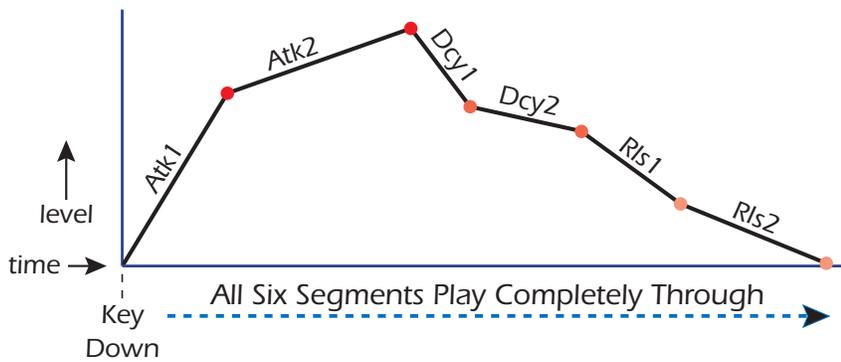
In Complete mode, the envelope generator continues through all six stages when a key is pressed, ignoring the status of key sustain. This feature allows you to create long evolving soundscapes with a quick tap of the key (one-shot mode), since normally the envelope would jump into the release stages as soon as the key was released. This mode is also well suited for enveloping percussive sounds such as piano and drums, or plucked strings which decay.

Complete mode can also be used to create more processor-efficient presets. On Emulator X, the voice stops playing when the Amp Envelope ends, and the computer resources used by that voice become available for other jobs. Because the envelope completes its cycle regardless of how long a key (or sustain pedal) is held, the computer resources are only used for a fixed amount of time, then returned. The performance increase can be dramatic!

- **Tip:** One-Shot mode.
 1. Select "Complete" mode.
 2. Set the envelope controls as shown below.
 3. Adjust Attack 2 time.



Envelope Complete Mode



Time-based and BPM Envelopes

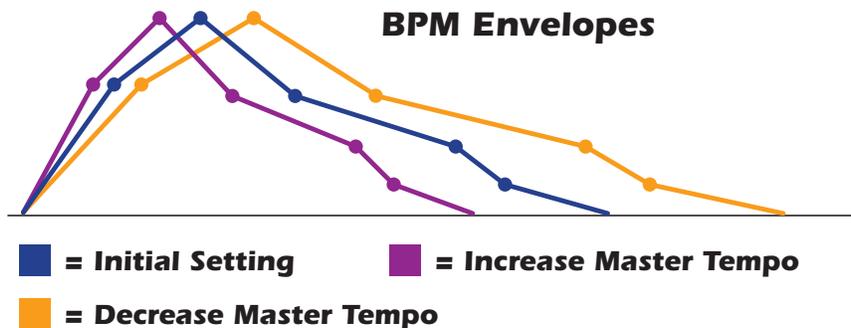
Envelope times can be time-based or BPM-based. Time-based envelopes have an absolute time setting, such as 1 second. BPM-based envelope times are divided down from the Master Tempo setting on the Multisetup page. The Master Tempo scales the time of the BPM-based envelope segments.

BPM times are displayed in musical time intervals where possible. For example, if you set the attack time to be 1/16, the attack time will last exactly one sixteenth-note at the master tempo.

BPM Envelopes

BPM envelopes are controlled by the Master Tempo (located in the Multisetup window). The Master Tempo rate scales the time of all the BPM envelope segments. The Master Tempo can also be derived from an external MIDI clock to synchronize the envelope times with external sequencer or arpeggiator tempo changes. See note->.

- ◆ **Note:** The master tempo can be set internally using the tempo control in the Multisetup window, or controlled from an external MIDI clock. Internal/External clock selection is located in the Preferences dialog box.



The chart below lists all the displayed musical note divisions when the envelope is set to BPM. The “Master Tempo Clock Pulses” column shows the exact number of clocks used by each note division based on the MIDI standard of 24 pulses per quarter note. The envelope segments will last for the exact note division shown. You can also set envelope times between the standard note divisions. These are displayed as integer numbers.

BPM Tempo Chart

BPM Tempo Rates (based on Master Tempo)	Displayed Note Division	Master Tempo Clock Pulses
16x whole note (quad long)	16/1	1536
8x dotted whole note	8/1d	1152
16x whole note triplet	16/17	1024
8x whole note (double long)	8/1	768
4x dotted whole note	4/1d	576
8x whole note triplet	8/1t	512
4x whole note (long)	4/1	384
2x dotted whole note	2/1d	288
4x whole note triplet	4/1t	256
2x whole note (breve)	2/1	192
dotted whole note	1/1d	144
2x whole note triplet	2/1t	128
whole note	1/1	96
dotted half note	1/2d	72
whole note triplet	1/1t	64
half note	1/2	48
dotted quarter note	1/4d	36
half note triplet	1/2t	32
quarter note	1/4	24
dotted 8th note	1/8d	18
quarter note triplet	1/4t	16
8th note	1/8	12
dotted 16th note	1/16d	9
8th note triplet	1/8t	8
16th note	1/16	6
dotted 32nd note	1/32d	4.5
16th note triplet	1/16t	4
32nd note	1/32	3
dotted 64nd note	1/64d	2.25
32nd note triplet	1/32t	2
64th note	1/64	1.5

LFOs & Lags



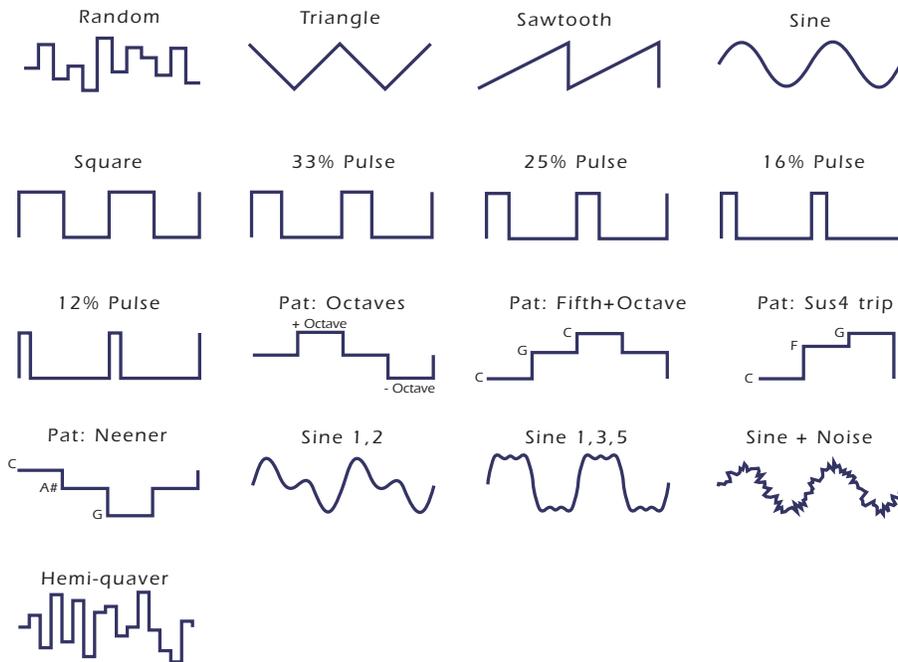
A Low Frequency Oscillator or LFO is simply a wave which repeats at a slow rate. LFOs are used to add animation to a sound. Emulator X has two multi-wave LFOs for each voice, with 17 different waveforms which are diagrammed below.

Frequency

Sets the initial speed of the LFO. The frequency range is 0.08 Hz (12.5 seconds for one cycle) to 18.01 Hz.

Note: References to musical intervals in the pattern LFO shapes are with the LFO routed to pitch and a PatchCord amount of +37.5.

See the Pitch Amount Chart on [page 158](#).



Delay

Sets the amount of time between playing a key and the onset of modulation. This can be used to simulate an effect often used by acoustic instrument players, where the vibrato is brought in only after the initial note pitch has been established. The delay range is adjustable from 0 seconds to 20 seconds.

Shape

Sets the LFO waveshape which determines the effect of the LFO. There are 16 LFO waveforms and a random, sample & hold type of wave which outputs random levels at the LFO rate.

Variation

Sets the amount of random variation on the LFO frequency each time a key is pressed, which is useful for ensemble effects, where each note played has a slightly different modulation rate. The higher the number, the greater the note to note variation in the LFO rate.

Sync

Key Sync or Free Running. With key sync selected, the LFO wave starts at the beginning of its cycle at each key depression. With Key Sync off, the LFO wave begins at a random point in its cycle on each key depression.

BPM

The BPM button allows the LFO frequency to be controlled by the Master Tempo (located in the Multisetup window). The Master Tempo can also be derived from an external MIDI clock to synchronize the envelope times with external sequencer or arpeggiator tempo changes. The chart below lists all the displayed musical note divisions when the LFO is set to BPM. The "Master Tempo Clock Pulses" column shows the exact number of clocks used by each note division based on the MIDI standard of 24 pulses per quarter note. The LFO cycle will last for the exact note division shown.

◆ The master tempo can be set internally using the tempo control in the Multisetup window, or controlled from an external MIDI clock. Internal/External clock selection is located in the Preferences dialog box.

BPM Tempo Rates (based on Master Tempo)	Displayed Note Division	Master Tempo Clock Pulses
8x whole note (double long)	8/1	768
4x dotted whole note	4/1d	576
8x whole note triplet	8/1t	512
4x whole note (long)	4/1	384
2x dotted whole note	2/1d	288
4x whole note triplet	4/1t	256
2x whole note (breve)	2/1	192
dotted whole note	1/1d	144
2x whole note triplet	2/1t	128
whole note	1/1	96
dotted half note	1/2d	72
whole note triplet	1/1t	64
half note	1/2	48
dotted quarter note	1/4d	36
half note triplet	1/2t	32
quarter note	1/4	24
dotted 8th note	1/8d	18
quarter note triplet	1/4t	16
8th note	1/8	12
dotted 16th note	1/16d	9
8th note triplet	1/8t	8
16th note	1/16	6
dotted 32nd note	1/32d	4.5
16th note triplet	1/16t	4
32nd note	1/32	3

Function Generators

The function generator is a multi-function modulation source, fully capable as a complex LFO, a sequencer, an arpeggiator, or an envelope generator, depending on the way it is utilized. Because of its amazing flexibility, the function generator is certain to be one of your favorite new toys in your sound design toolbox. Because you can never have too much of a good thing, each voice has three independent function generators!



The function generator outputs a stepped (or smooth) sequence of values and a gate/trigger for each step. Its most obvious function is a step-sequencer (when controlling pitch), or as a complex event generator for the filter or amplifier. Several key sync and direction modes allow you to create interesting variations and complex levels of control. The gate and trigger outputs can, trigger envelope generators, retrigger samples or LFOs, accent certain notes, change tempo, and trigger other events, which we'll describe in detail later on.

Function Display and Grid

The function display is a bar graph, with an additional line of rectangular cells. The bar graph displays the function levels associated with each step, while the rectangles display a pattern used to generate Trigger and Gate modulation signals.

The display shows all 64 steps, with all steps to the left of the End Step drawn in color; the steps to the right of the End Step (if any) are drawn in gray.

You can edit the function either by dragging across the display, or by dragging individual steps. A pop-up tool tip shows the step number and value.

There are three key commands to make curve drawing easier. Hold the appropriate key while moving the cursor.

- **Ctrl-key**..... Constrains to vertical dragging only. Useful for setting levels.
- **Shift-key**.... All levels are set to the same value, resulting in a horizontal line. This also works for the trigger outputs, turning them on or off.
- **Alt-key** Values are randomized when this key is held down while dragging. The trigger outputs, as well as the levels can be randomized. The values are interpolated when dragging quickly, which can result in interesting random curves.

The **Single Step Controls** allow you to precisely set the value and trigger status of each individual step. This method is useful for entering a prearranged composition.

The Grid controls determine the spacing of a set of vertical grid lines that are overlaid on the function display (see the following page).

The grid options are listed below:

- **None**
- **Semitone**..... One line for each unit (semi tones).
- **Octaves** One line for each 12 units.
- **Major Triad**..... Lines correspond to a major triad, repeated over the full range.
- **Minor Triad** Lines correspond to a minor triad, repeated over the full range.
- **Major Scale**..... Lines correspond to a major scale, repeated over the full range.
- **Minor Scale** Lines correspond to a minor scale, repeated over the full range.

The musical intervals on the grid are helpful when the function generator is modulating **Pitch** and the Cord amount is set to **100%**.

The **Grid Button** turns on Snap-to-Grid, which forces the function values to snap to the grid lines.

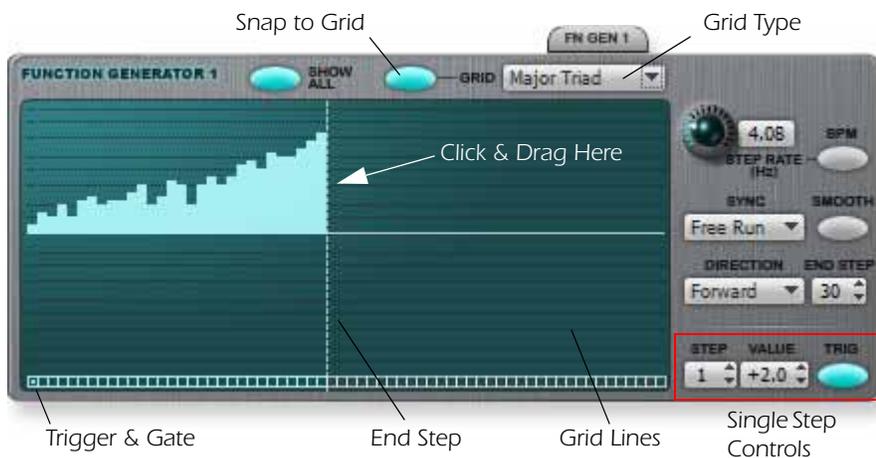
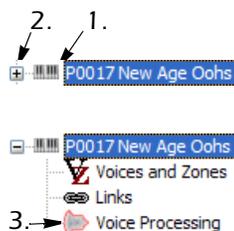
The **Show All** button allows you view all 64 steps (when on), or only shows the steps before the End Step. This feature is very useful with shorter sequences as it lets you “zoom in” to the area of interest.

► **Using the Function Generator as a Sequencer / Arpeggiator**

Using the Function Generator as a pitch controller is the most obvious application. It’s also lots of fun and the easiest way to explore the various controls.

Set up

1. Select **P0017 New-Age Oohs** from the Tree by clicking on the little keyboard.
2. Next click on the (+) symbol to open the preset modules.
3. Click on the **Voice Processing icon** . The Voice Processing module appears.
4. Go to PatchCord #18 and patch “**Func Gen 1 +**” to “**Pitch**” with an amount of 100%. 
5. Now click the **Func Gen 1** tab. The screen below appears.
6. Turn on “Snap to Grid” by pressing the **Grid** button.



7. Select **Major Scale** as the Grid Type.

Modify the Sequence & Play

8. Click in the graphic display and begin horizontally dragging the mouse. Notice how the bars snap to the Grid lines.
9. **Play a key.** Play a chord. Play an arpeggio.
10. Go ahead and adjust the sequence until you get something you like. The function generator editing controls are described on [page 139](#). They're easy to use once you get the hang of it.
11. Click and drag the **End Step** line to shorten or lengthen the sequence.
12. Change the **Step Rate** to change the speed of the sequence. (The BPM button locks the step rate to a note division of the Master Tempo.)
13. Now change the **Direction**. There are six different direction options described on [page 143](#).
14. Change the **Grid Type** and create a new sequence
15. Play around with the controls as much as you like. The function generators are a lot of fun and have many, many applications. You've just created a type of programmable arpeggiator. Later on, we'll describe how to add rests, accents, change the timing of notes and do a few other tricks.

■ **Tip:** Turn "Show All" off to zoom in and only display the steps in your sequence.

Function Generator Main Controls

Control	Function
Step Rate	Sets the step frequency. This value controls the frequency at which individual steps occur, not the overall rate of pattern repetition. The frequency range for each step is 0.081 Hz to 18.147 Hz.
BPM	Changes the Frequency control to a clock divider control to change steps on quarter notes, eighth notes, etc. See the BPM chart on page 137 for the complete list of possible clock divisors.
End Step	This number sets the last played step from 1 to 64. This field can be modulated between 1 and 64. The end step can also be changed by simply dragging the end step line in the graphic display. A handy pop-up tool tip shows the step number and the value.
Sync	There are three sync modes: Key , Freerun , and Channel . <ul style="list-style-type: none"> ▪ Key Sync: The function generator begins at the first step for each new key played. Each key follows its own pattern. ▪ Freerun: The pattern begins at a random step for each new key played. Each key follows its own pattern. ▪ Channel Sync: The pattern begins with the first key down on the assigned MIDI channel and continues until no keys are held down on the MIDI channel. All keys on the channel follow the same pattern.
Smooth	When on, linear interpolation is applied between steps to provide a smooth output instead of a stepped function.

Function Generator Main Controls

Control	Function
Direction	<p>This control determines the order in which the steps are played.</p> <ul style="list-style-type: none"> ▪ Forward: Normal playback, loops back to the beginning when the end is reached. ▪ Reverse: Starts at the last step and counts back, looping from beginning to end. ▪ Pendulum (back and forth): Starts at the first step, precedes to the last, and then reverses. Each end point is played once. ▪ Random: The next step is chosen at random. ▪ Brownian: (“drunkards walk”) The next step is chosen at random, but it is always the next or previous step in the function. In other words, the next step is always adjacent to the current step, but the direction of motion is chosen at random. Motion will bounce off the endpoints; when an endpoint is reached, the next step is always the one that is inward in the function. ▪ One-shot: Forward, plays once and then stops at the end.

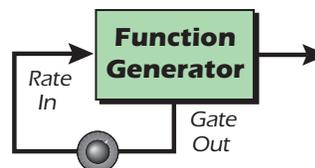
Modulation Destinations

The following modulation destinations are available for each function generator:

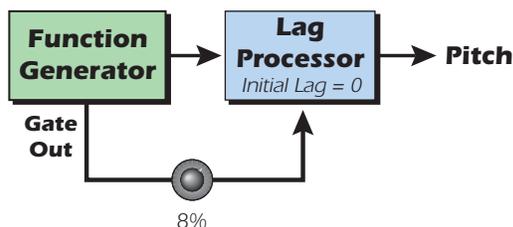
Rate	<p>Positive modulation increases the initial rate. Negative values decrease the initial rate.</p> <ul style="list-style-type: none"> A modulation value of +1.0 (Cord: +100%) will result in the maximum step rate (18.147 Hz) from any initial setting. A modulation value of -1.0 (Cord: -100%) will result in the minimum step rate (0.081 Hz) from any initial setting.
Direction	<p>Changes the direction (forward <-> backward). This is evaluated at each step. Modulation of direction is only valid when the initial direction type is set to forward or backward. A positive modulation value (>0.0) will result in a forward direction. A negative modulation value (<0.0) will result in a reverse direction.</p>
Retrigger	<p>Restarts the function generator when the modulation source crosses zero in the positive direction.</p>
Length	<p>Modulating the Length changes the end step as follows:</p> <ol style="list-style-type: none"> Modulation adds to the user specified end step. <ul style="list-style-type: none"> A mod value of +1.0 (Cord: +100%) adds 64 steps to the end step (clipped at 64). A mod of -1.0 (Cord: -100%) adds -64 steps to the end step (clipped at 1). If the current step exceeds the sum of the End Step and the Modulation Amount, the function generator will: <ul style="list-style-type: none"> Return to the starting step for the single direction types such as forward, backward, etc. in a looping type fashion. Reverse direction in the case of the pendulum. Stop in the case of the one shot direction type.

Function Generator Ideas

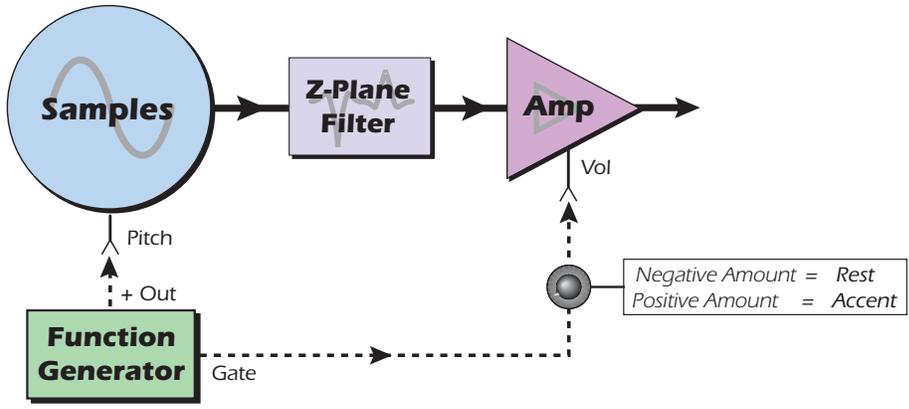
- **Speed Control** - You can slow down or speed up certain notes by routing the Function Generator Gate back to control its own Rate. Positive Cord values speed up the gated notes; negative Cord values slow down gated notes.



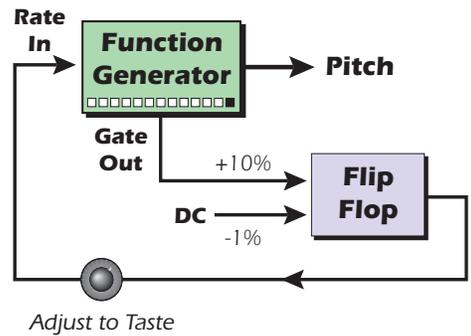
- **Glide Control** - Only notes that have the Gate turned on will glide using this patch. This technique works well for all kinds of music including synth bass lines and adds another element of interest to the sound.



- **Rests and Accents** - You can insert rests and accents onto specific notes using this simple patch. To make a rest, set the amp cord amount negative (turning the amp off). For an accented note, set the cord amount positive (making the amp louder).

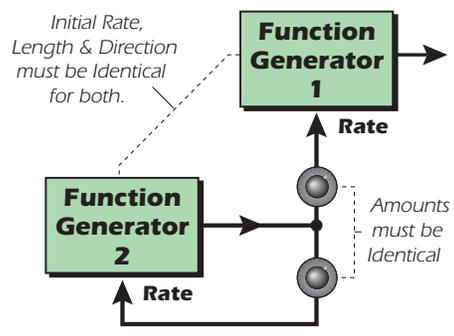


- **Two-Speed Sequence** - This sequence changes rate every other time it plays through. The flip-flop sets and resets every other time it is pulsed with the function generator gate. The Cord amount from the flip-flop controls the amount of speed variation. *(Note: The -DC Cord is necessary because the flip-flop input needs to go positive from a negative value.)*

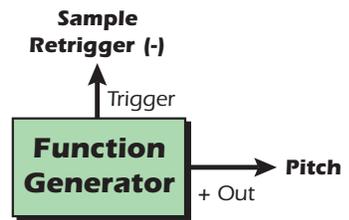


- **Multi-Speed Sequence** - If you want to control the timing for each step in a sequence, two function generators can be used. Function Generator 1 controls the pitch and Function Generator 2 controls the length of each note.

Two function generators will remain in sync if their initial rate, length and direction are the same, and if the modulation is identical for both.



- **Sample Re-triggering** - The Trigger Out of the Function Generator is ideal for re-triggering samples. In fact, the trigger out can be used just for sample re-triggering without using any pitch control. Use Sample Retrigger (-) to trigger on the note. (Remember, samples to be retriggered must have a loop.)



Filter Controls



You can choose from 55 different filter types or choose “No Filter”, which bypasses the filter section.

Most of the filters have two parameters: Frequency (or Morph) and Resonance (Q, Gain, Body Size). These two parameters can be continuously varied during the note (choose realtime resonance).

The frequency response curve of the filter is accurately shown in the display as you change the initial filter settings. Frequency is shown on the horizontal axis, with amplitude on the vertical axis.

High values of Q or resonance amplify frequencies near the cutoff or center frequency. In a swept EQ filter, gain controls the amount of boost or cut. In a phaser or flanger, resonance determines the depth of the effect. In the vocal filters, body size determines the apparent size of the mouth cavity.

You will notice an increase in voice count when the filters are bypassed as the filters steal cycles from your computer’s CPU. 12th order filters use more CPU and therefore decrease the maximum voice count.

Filter Descriptions

Filter Name	Order	Type	Description
2-pole Lowpass	02	LPF	Typical OB type low-pass filter with a shallow 12 dB/octave slope.
4-pole Lowpass	04	LPF	4-pole low-pass filter, the standard filter on classic analog synths. 24 dB/octave rolloff.
6-pole Lowpass	06	LPF	6-pole low-pass filter which has a steeper slope than a 4-pole low-pass filter. 36 dB/octave rolloff!
2-pole Highpass	02	HPF	2-pole high-pass filter. 12 dB/octave slope.

Filter Types

BPF	Band-pass filter
EQ+	EQ boost filter
EQ-	EQ cut filter
FLG	Flanger filter
HPF	High-pass filter
LPF	Low-pass filter
PHA	Phaser filter
PROG	Programmable
REZ	Highly Resonant
SFX	Special Effect
VOW	Vowel / formant
WOW	Wah-Wah pedal

Filter Name	Order	Type	Description
4-pole Highpass	04	HPF	Classic 4-pole high-pass filter. Cutoff sweep progressively cuts 4th Order High-pass.
2-pole Bandpass	02	BPF	Band-pass filter with 6 dB/octave rolloff on either side of the passband and Q control.
4-pole Bandpass	04	BPF	Band-pass filter with 12 dB/octave rolloff on either side of the passband and Q control.
Contrary Band pass	06	BPF	A novel band-pass filter where the frequency peaks and dips midway in the frequency range.
SweptEQ 1 octave	06	EQ+	Parametric filter with 24 dB of boost or cut and a one octave bandwidth.
Swept EQ 2->1 octave	06	EQ+	Parametric filter with 24 dB of boost or cut. The bandwidth of the filter is two octaves wide at the low end of the audio spectrum, gradually changing to one octave wide at the upper end.
Swept EQ 3->1 octave	06	EQ+	Parametric filter with 24 dB of boost or cut. The bandwidth of the filter is three octaves wide at the low end of the audio spectrum, gradually changing to one octave wide at the upper end.
Phaser 1	06	PHA	Recreates a comb filter effect typical of phase shifters. Frequency moves position of notches. Q varies the depth of the notches.
Phaser 2	06	PHA	Comb filter with slightly different notch frequency moving the frequency of notches. Q varies the depth of the notches.
FlangerLite	06	FLG	Contains three notches. Frequency moves frequency and spacing of notches. Q increases flanging depth.
Vocal Aah-Ay-Eeh	06	VOW	Vowel formant filter which sweeps from "Ah" sound, through "Ay" sound to "Ee" sound at maximum frequency setting. Q varies the apparent size of the mouth cavity.
Vocal Ooh-Aah	06	VOW	Vowel formant filter which sweeps from "Oo" sound, through "Oh" sound to "Ah" sound at maximum frequency setting. Q varies the apparent size of mouth cavity.
Dual EQ Morph	06	PROG	Please refer to the detailed description on page 151 .
Dual EQ + Lowpass Morph	06	PROG	Please refer to the detailed description on page 153 .
Dual EQ Morph + Expression	06	PROG	Please refer to the detailed description on page 152 .
Peak/Shelf Morph	06	PROG	Please refer to the detailed description on page 153 .
Morph Designer	2 - 12	PROG	Please refer to the detailed description on page 149 .

Filter Name	Order	Type	Description
Ace of Bass	12	EQ+	Bass-boost to bass-cut morph
MegaSweepz	12	LPF	"Loud" LPF with a hard Q. Tweeters beware!
EarlyRizer	12	LPF	Classic analog sweeping with hot Q and Low-end.
Millennium	12	LPF	Aggressive low-pass filter. Q gives you a variety of spiky tonal peaks.
MeatyGizmo	12	REZ	Filter inverts at mid-Q.
KlubKlassik	12	LPF	Responsive low-pass filter sweep with a wide spectrum of Q sounds
BassBox-303	12	LPF	Pumped up lows with TB-like squelchy Q factor.
FuzziFace	12	DST	Nasty clipped distortion. Q functions as mid-frequency tone control.
DeadRinger	12	REZ	Permanent "Ringing" Q response. Many Q variations.
TB-OrNot-TB	12	EQ+	Great Bassline "Processor."
Ooh-To-Eee	12	VOW	Oooh to Eeee formant morph.
BolandBass	12	EQ+	Constant bass boost with mid-tone Q control.
MultiQVox	12	VOW	Multi-Formant, Map Q To velocity.
TalkingHedz	12	VOW	"Oui" morphing filter. Q adds peaks.
ZoomPeaks	12	REZ	High resonance nasal filter.
DJAlkaline	12	EQ+	Band accentuating filter, Q shifts "ring" frequency.
BassTracer	12	EQ+	Low Q boosts bass. Try sawtooth or square waveform with Q set to 115.
RogueHertz	12	EQ+	Bass with mid-range boost and smooth Q. Sweep cutoff with Q at 127.
RazorBlades	12	EQ-	Cuts a series of frequency bands. Q selects different bands.
RadioCraze	12	EQ-	Band limited for a cheap radio-like EQ
Eeh-To-Aah	12	VOW	"E" to "Ah" formant movement. Q accentuates "peakiness."
UbuOrator	12	VOW	Aah-Uuh vowel with no Q. Raise Q for throaty vocals.
DeepBouche	12	VOW	French vowels! "Ou-Est" vowel at low Q.
FreakShifta	12	PHA	Phasey movement. Try major 6 interval and maximum Q.
CruzPusher	12	PHA	Accentuates harmonics at high Q. Try it with a sawtooth LFO.
AngelzHairz	12	FLG	Smooth sweep flanger. Good with vox waves. e.g. 1094, Q =60
DreamWeava	12	FLG	Directional Flanger. Poles shift down at low Q and up at high Q.

6 - Voice Processing
 Amplifier, Filter & Auxiliary Envelopes

Filter Name	Order	Type	Description
AcidRavage	12	REZ	Great analog Q response. Wide tonal range. Try it with a sawtooth LFO.
BassOMatic	12	REZ	Low boost for basslines. Q goes to distortion at the maximum level.
LucifersQ	12	REZ	Violent mid Q filter! Take care with Q values 40-90.
ToothComb	12	REZ	Highly resonant harmonic peaks shift in unison. Try mid Q.
EarBender	12	WAH	Midway between wah & vowel. Strong mid-boost. Nasty at high Q settings.
KlangKling	12	SFX	Ringing Flange filter. Q "tunes" the ring frequency.

Programmable Morphing Filters

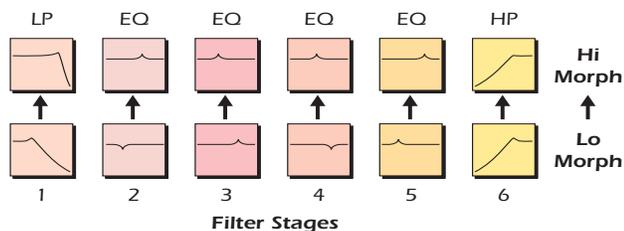
There are five user-programmable morphing filters which have the ability to change their response over time. You can design your own custom filters that morph between filter types.

Morph Designer	A supremely programmable filter with up to six filter sections.
Dual EQ Morph	A 2-frame morphing filter with two EQ sections and controllable gain.
2EQ Morph + Expression	This filter contains two EQ sections and a separately controllable lowpass filter.
2EQ Morph + Lowpass Morph	This filter has two EQ sections and lowpass filter tied to the Morph parameter.
Peak/Shelf Morph	This is a 2-frame morphing filter with independent control over frequency, shelving and peak for each filter frame.

Take the time to learn to use these powerful filters because they sound great. The frequency response graphs are especially helpful to understand the action of these complex filters since the control functions are slightly different for each filter.

Morph Designer

The Morph Designer is the ultimate in synthesizer filters with up to six completely programmable filter sections (lowpass, highpass or EQ) and the ability to morph between two completely different frame settings of the six sections.



Any of the six filter sections can be designated as either a lowpass, highpass, or EQ filter. You can set the filter frequency and Q (or Gain in the case of EQ) for both the low and high positions of the Morph control. When the Morph control is changed, all the filter controls interpolate between the settings you programmed. Twenty-four knobs have been reduced to just two extremely powerful controls, Morph and Gain (offset).



EQ Stage Gain Controls

- The small Gain knobs control Gain (-24dB to +24 dB)
- The big Gain Wheel is Gain Offset (-24dB to +24dB)
- For any given section, the two Gain controls add (clipping at 0 or 100%)

HP/LP Stage Gain Controls

- The small Gain knobs control Q (0 to 100%)
- The big Q Wheel is Q Offset (-50% to +50%)
- For any given section, the two Q controls add (clipping at 0 or 100%)

■ **Tip:** Set the Gain/Q Wheel in the **center** while programming your Morph Designer filter, and use the Morph control to sweep the filter.

► **Explore the Morph Designer:**

You can create great sounding filters with just a few filter sections.

1. Select any preset with lots of harmonic content.
2. Go into the Voice Processing window, select all Groups and select **Morph Designer** as the filter type.
3. Turn **MIDI Controller A** all the way down on your MIDI keyboard.

First Filter Section

4. Select **LP** as the shape for filter Stage 1.
5. Turn the **Morph Wheel** all the way down. (*Select the Wheel, then press Home.*)
6. Set the **Gain Wheel** to its center position. (*Alt+click the mouse on the control.*)
7. Set the **Frequency** and **Gain** controls of the **Lo Morph** all the way down.
8. Turn the **Morph Wheel** all the way up. (*Select the Wheel, then press End.*)
9. Set the **Frequency** and **Gain** controls of the **Hi Morph** to their maximum settings.
10. Now play the keyboard and turn the Morph wheel. You have just created the classic synthesizer lowpass filter response, but in this case the Q turns up automatically when you change the filter frequency. If you turn up the Q Wheel, you get even more Q. (*The Q control adds to the Lo and Hi Q settings.*)

Add Another Section

11. Select **Stage 2** and this time choose **EQ** as the filter type.
 12. Turn the **Morph Wheel** down (you can leave the Gain Wheel centered).
 13. Set the **Lo Morph Frequency** to the minimum setting.
 14. Set the **Low Morph Gain** to zero. (*Alt+click the mouse on the control.*)
 15. Turn the **Morph Wheel** to the maximum setting.
 16. Set the small **Hi Morph** controls so that another little resonance peak appears next to the one from the lowpass filter. The filter response display should look something like this:
- 
17. Now play the keyboard and turn the **Morph Wheel**. The dual resonance peaks come in as the filter frequency (morph) is turned up, and they sound much more interesting than a simple lowpass.
 18. Go ahead and adjust the controls some more, noting the interaction between the Lo/Hi Morph controls and the Morph Wheel. It's really pretty simple. The Morph Wheel interpolates between the Lo and Hi settings of the filter section.

Add Another Section

19. When you're ready go ahead and add yet another filter section. You're on your own now so choose any filter type you want. Using just three of the six sections you can create some truly great sounding filters.

Morph Designer Controls

Control	Function
Morph (Frequency)	The Morph control actually morphs between the Low Morph and High Morph Frames of the filter. The Morph control is a modulation destination.
Gain/Q (Resonance)	Increases or decreases the gain or Q of all filter sections simultaneously. This adds to the settings made in the Lo and Hi Morph controls. The Resonance control is a modulation destination.
Section	Selects which filter section you are currently editing.
Shape	Sets the filter shape for the current section being edited. <ul style="list-style-type: none"> ▪ Off.....Disables the filter section ▪ EQ..... Peak or Dip response ▪ Lowpass....Lowpass response with Q ▪ Highpass..Highpass response with Q
Lo Morph Frequency	Sets the center frequency of the current section when the Morph control is at minimum.
Lo Morph Gain/Q	Sets the Gain (if EQ) or Q (if lowpass or highpass) of the current section when the Morph control is at minimum.
Hi Morph Frequency	Sets the center frequency of the current section when the Morph control is at maximum.
Hi Morph Gain/Q	Sets the Gain (if EQ) or Q (if lowpass or highpass) of the current section when the Morph control is at maximum.

Dual EQ Morph



This is a programmable 2-frame Morphing filter with two EQ sections. As the Morph value is increased, the filter interpolates from the low to high settings. The Gain of each section remains constant during the Morph, but can be scaled at note-on time by controlling the Initial Gain parameter.

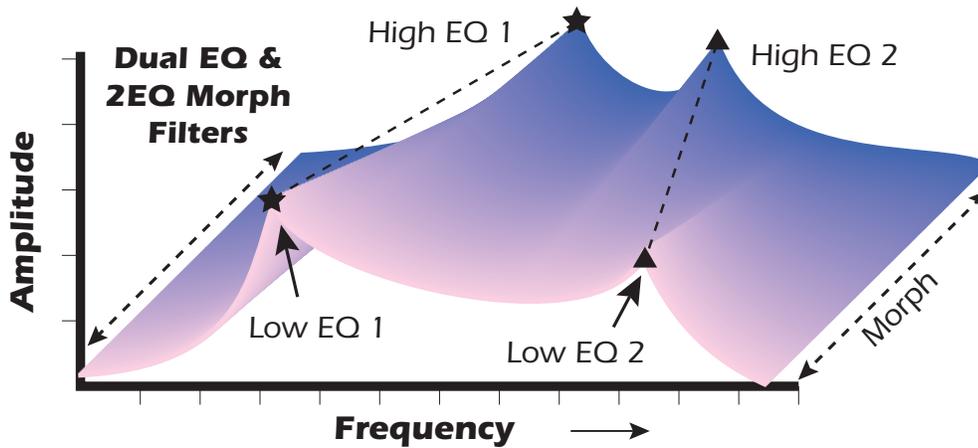
This filter can be used to create your own vocal formant filters.
(Try the settings shown and modulate with the + filter envelope or +LFO.)

Dual EQ Morph + Expression



This filter also has two EQ sections and a lowpass filter with independent control over the lowpass filter frequency (Expression). This allows you to use velocity (for example) to control expressive timbre via the lowpass filter, while the Morph parameter is modulated for another effect. As the Morph value is increased, the filter interpolates from the low to high settings. The gain parameter for each EQ section stays constant during the morph.

- ◆ Morph = Filt. Freq.
Expression = Filt. Res.



As the Morph value is increased, the filter interpolates from the Low to High Frequency setti

2EQ + Lowpass Morph



This unusual filter has two EQ sections and a lowpass filter whose Fc is also controlled by the Morph parameter. The Lowpass filter is completely open at a setting of 255. As the Morph value is increased, the filter interpolates from the low to high settings and the lowpass filter allows more high frequencies to pass. The Q of the lowpass filter has an initial setting and can also be modulated realtime using the Filter Resonance parameter. The gain parameter for each EQ section stays constant during the morph.

- ◆ $F_c/\text{Morph} = \text{Filt. Freq.}$
 $\text{LPF Q} = \text{Filt. Res.}$

Peak/Shelf Morph



This 2-frame morphing filter offers independent control over frequency, shelving and peak for each of the two filter "frames". The Peak parameter can also be modulated in realtime by controlling the Filter Resonance parameter.

- ◆ $\text{Morph} = \text{Filt. Freq.}$
 $\text{Peak} = \text{Filt. Resonance}$

When the Shelf parameter is negative, the filter will have a low shelving response. When the Shelf parameter is positive, the filter will assume a high shelving response. With Shelf at zero, the filter assumes a peak filter response.

Note that the three controls of the two morphing frames are grouped in this filter and all three controls are morphed.

Amplifier Controls



These parameters set the overall volume and pan settings, as well as the output channels for the selected voice(s).

The voice effect sends allow you to route the voice output to the main and aux send busses.

The Classic Response button matches the amplifier response to that of the Emulator IV.

Volume

Sets the initial volume of the voice from -96 dB (off) to +10 dB. This is the same volume control as shown on the Mix/Tune tab of Voices & Zones ([page 96](#)).

Lower the volume of the Amplifier when modulating Amp Volume. (You won't hear any modulation if the volume is already turned all the way up.)

The default setting is 0 dB which allows a bit of headroom so that other modulators can be added in without clipping. Use this volume control to balance the voice output against others. If you hear clipping, reduce volume setting.

Panning

Sets the initial stereo pan position of the voice. Subsequent modulation is added to or subtracted from this value. This is the same pan control as shown on the Mix/Tune tab of Voices & Zones ([page 96](#)).

Amp Envelope Dynamic Range

Sets the maximum amount of attenuation from the amplifier envelope generator. Variable from -96 dB to -48 dB. With longer attack times, reducing the envelope depth (towards -48 dB) may provide a more natural-sounding volume envelope.

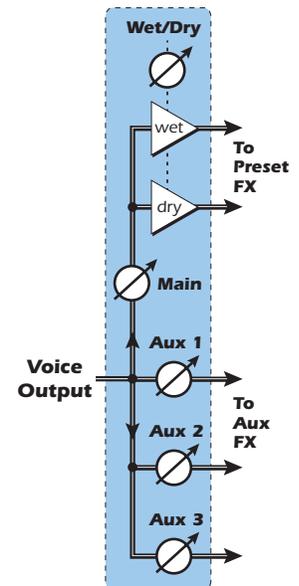
Response

Selects between the "Classic" amplitude response and "Emulator X" response. Choose Classic response ONLY if you're authoring for hardware Emulators. This button should be Off if you're creating sounds for the Emulator X.

FX Wet/Dry

This control sets the mixture of Wet/Dry signal that will be sent from the Voice Main output to the Preset FX processors. See the diagram at right. ->

A setting of 0 means that no signal from the Main output of the voice will be sent to the Preset FX. A setting of 100 means that all of the Main voice output will be sent through the Preset FX.

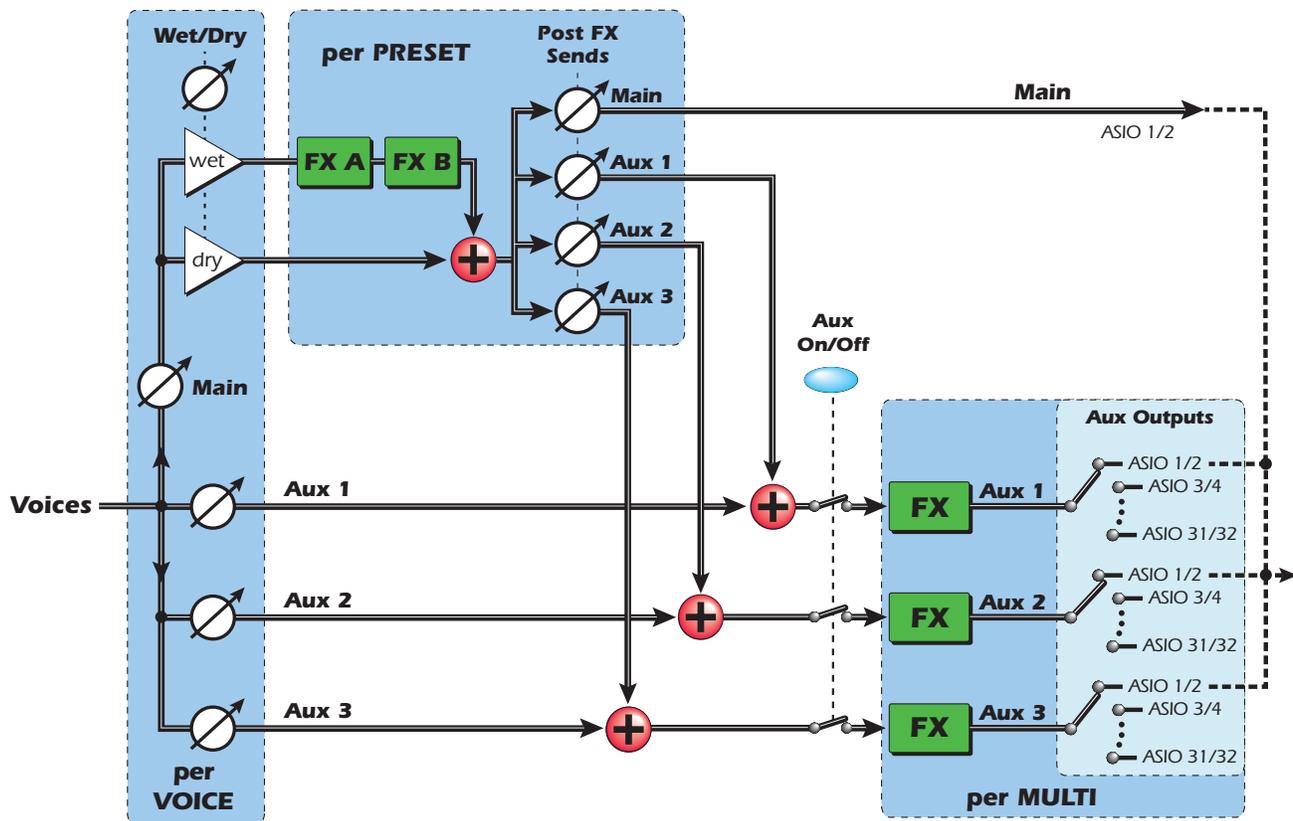


Voice Effect Sends

Each voice has one Main Output level control which feeds a Wet/Dry mixer, and three Aux Send level controls. The Wet/Dry outputs from all voices in the preset are summed into the Preset FX. The output of the Preset FX section can be mixed into the Main and Aux 1-3 outputs. The Main Output of the Preset FX goes directly to the channel's main stereo output. The Aux Outputs of the Preset FX are summed together with the Aux Output signals from all the voices.

All the Aux Sends 1-3 from all voices and channels are summed with Aux Outputs 1-3 from the Preset FX and then feed the Aux FX processors located on the Multisetup page. [See page 73](#). The outputs of the Aux FX processors (which may be bypassed) feed the Aux Output section, where they can be directed to any available ASIO output.

The Aux Output On/Off button to the right of each channel main output allows you to turn the Aux Sends on or off for that channel so you can hear the channel with or without effects.

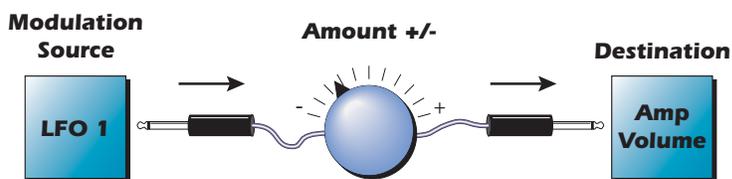
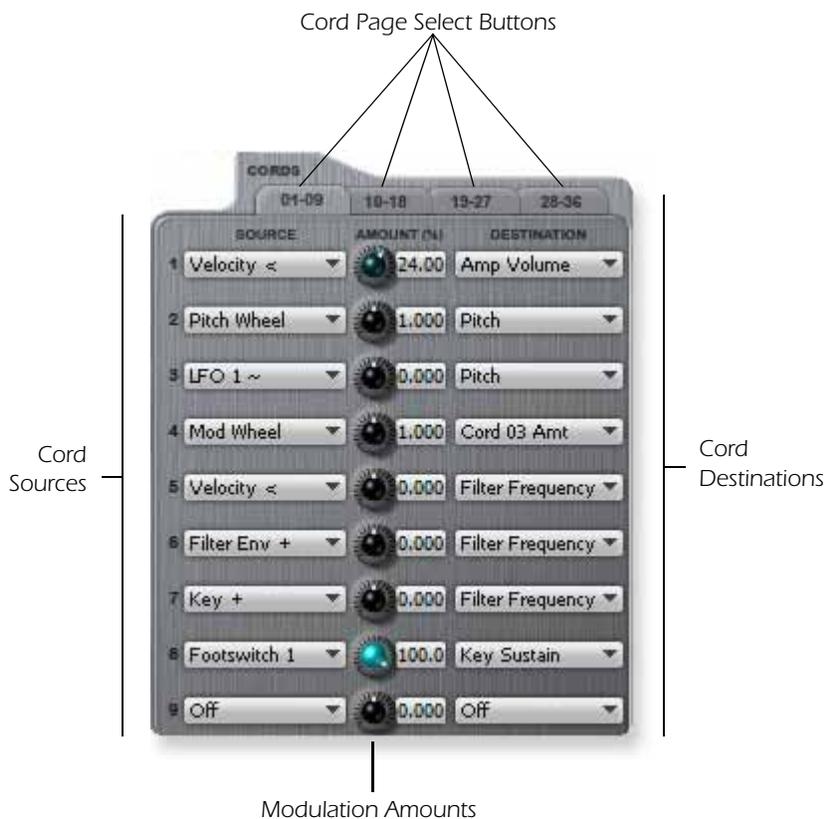


The Voice Effect Send levels and the Wet/Dry mix can be controlled using the voice modulation cords. [See "Send Aux 1-3" on page 159](#).

See also ["Controlling the Preset Effect Amount from the Voice" on page 239](#).

Modulation Cords

Each voice contains 36 Cords which are used to route modulation sources to destinations and control the amount of modulation. There are only three permanently connected modulation connections: amp envelope-to-amplifier, pitch wheel-to-pitch, and key-to-pitch. All other modulation routings must be connected with a Cord.



Tremolo is created by routing an LFO to the Amp Volume using a Cord.

Modulation Sources

Source	Description
Crossfade Random	Generates the same random value for all voices in a preset. This source is intended to be used for crossfading voices.
Key (+, ~)	Key Tracking - This value increases as you play up the keyboard.
Velocity (+, ~, <)	Key Velocity - This value increases as the key is played harder.
Release Velocity	This value increases the slower you release the key

Modulation Sources

Source	Description
Gate	This value goes to full value as long as a key is depressed.
Pitch Wheel	MIDI data from your keyboard's pitch bender.
Mod Wheel	MIDI data from your keyboard's modulation wheel or lever.
Pressure	Aftertouch MIDI data (channel pressure)
Pedal	Footpedal MIDI data (normally controller #4)
MIDI A - P	16 assignable MIDI continuous controllers. See page 28.
Footswitch 1 & 2	MIDI footswitch data (normally controllers 64 & 65)
F-F Footswitch 1&2	MIDI footswitch inputs which latch on or off with each press.
T'switch	MIDI "Thumb-switch" input. General switch controller.
T'switch FF	MIDI flip-flop thumb-switch input. Latching switch controller.
MIDI Volume	MIDI Continuous Controller data on channel #7
MIDI Pan	MIDI Continuous Controller data on channel #10
Expression	MIDI Expression Controller data (MSB) on channel #11
Key Glide	A smoothly changing control source based on the Glide Rate and the interval between the last two notes played.
Key CC Window	Keyboard keys output as a modulation source, used in conjunction with the "Key CC Window". See page 85.
Amp Envelope	Amplifier Envelope Generator (+, ~, <). See page 159.
Filter Envelope	Filter Envelope Generator (+, ~, <). See page 159.
Aux Envelope	Auxiliary Envelope Generator (+, ~, <). See page 159.
LFO 1 & 2 (+, ~)	Low Frequency Oscillators 1 & 2 (+, ~,)
Func Generators 1-3	Function Generator 1, 2 & 3 outputs (+, ~, <). See page 139.
Func Gen Gate & Trig	Function Generator Gate and Trigger outputs. See page 143.
White Noise	A continuously varying random control source for effects.
Pink Noise	A lower frequency random control source.
KRandom 1 & 2	These sources generate different random values for each voice. The values are chosen at key-on time and remain constant.
Lag 0 & 1 in	Lag Processors can also be used as summing amps. See page 138.
Lag Processor 1 & 2	Lag processors slow down rapid changes. See page 138.
Channel Lags 1 & 2	Lag processors that operate at the channel level. See page 85.
Channel Ramp	Slope generator that is triggered by the first key pressed. See page 86.
Poly Key Timer	Outputs a value based on the time a key is held down. See page 86.
Clock Divisors	MIDI clock is divided to the specified value and used as a modulation source. See page 304.
DC Offset	A fixed value that can be scaled by the Cord to offset a value.

Modulation Sources

Source	Description
Summing Amp	The Summing Amp is useful for mixing several modulation signals so they can be controlled from a single Cord amount. See page 307.
Switch	The switch outputs full value (100) when the input is greater than 0. (The output value can be scaled by a Cord.) See page 307.
Absolute Value	This processor inverts negative values and leaves positive values untouched so that only positive values are output. See page 307.
Diode	This processor allows only positive values to pass. Negative values are blocked. See page 307.
Flip-Flop	The output of this processor alternates between full value and zero each time the input goes positive from zero or a negative value. If an LFO were connected to the input, the output would be a square wave of half the input frequency. See page 307.
Quantizer	Smoothly changing signals are transformed into discrete steps. The value of the input Cord determines the number of steps. The value of the output Cord determines the size of the steps. (Hint: Gain 4x can increase number of steps or step size.) See page 307.
Gain 4x	This processor amplifies the modulation input by a factor of 4.

Pitch Amount Chart

In order to obtain specific pitches when routing LFOs, envelopes and controllers through the PatchCords, use the following chart.

Semitone	Cord Amt	Semitone	Cord Amt	Semitone	Cord Amt
1	3.125	9	28.125	17	53.125
2	6.25	10	31.25	18	56.25
3	9.375	11	34.375	19	59.375
4	12.5	12	37.5	10	62.50
5	15.625	13	40.625	21	65.625
6	18.75	14	43.75	22	68.75
7	21.875	15	46.875	23	71.875
8	25.00	16	50.00	24	75.00

Modulation Destinations

Destination	Description
Key Sustain	Holds the envelope generators at the sustain point as long as this value is above zero.
Fine Pitch	Fine pitch modulation with a maximum range of 1 semitone.
Pitch	Full range pitch modulation.
Glide	Controls the portamento (glide) time.
Chorus Amount	Controls the amount of voice chorus (doubling).
Chorus Position ITD	Controls the Inter-aural Time Delay of voice chorus. See page 130.
Sample Start	Controls the sample start offset point. See page 131.
Sample Loop	Controls the sample loop length. See page 161.
Sample Retrigger (+)	A positive-going, zero-crossing value retriggers the sample from the sample start. Note: A sample must be looped in order to retrigger properly. For unlooped sounds, create a silent loop at the end of the sample.
Sample Retrigger (-)	A negative-going, zero-crossing value retriggers the sample from the sample start.
Loop Select (Continuous)	Loop selection that plays through to the next selected loop. Only backward jumping is allowed. See page 219.
Loop Select (Jump)	Loop selection that jumps to the selected loop at the end of the currently playing loop. Forward and backward jumping is allowed. See page 221.
Filter Frequency	Controls the frequency (or morph) of the filter.
Realtime Resonance	Realtime control of filter resonance (Q, gain, body size).
Filter Resonance	Filter resonance control which is set at note-on time only.
Amp Volume	Controls the volume of the voice.
Amp Pan	Controls the left/right positioning of the voice.
Amp Crossfade	Controls the RT crossfade of voices so programmed. See page 105.
Send Main	Controls the level of the voice sent to the Preset FX and the Main Output. See page 232.
Send Aux 1-3	Controls the level of the voice sent to the Aux Send 1, 2 or 3 FX processors. See page 232.
Wet/Dry Mix	Controls the Wet/Dry mix sent to the Preset FX. See page 232.
Amp Envelope Rates	Positive values decrease the rates of all amp envelope stages.
Amp Env. Attack	Positive values decrease the attack rate of the amp envelope.
Amp Env. Decay	Positive values decrease the decay rate of the amp envelope.
Amp Env. Release	Positive values decrease the release rate of the amp envelope.
Filter Envelope Rates	Positive values decrease the rates of all filter envelope stages.
Filt Env Atk, Dcy, Rel	Same as for Amplifier. See above.
Filter Env. Trigger	A positive-going, zero-crossing value retriggers the envelope.
Aux. Envelope Rates	Positive values decrease the rates of all Aux. envelope stages.
Aux Env Atk, Dcy, Rel	Same as for Amplifier Envelope parameters. See above.
Aux Env. Trigger	A positive-going, zero-crossing value retriggers the envelope.
LFO Rate (1 & 2)	Positive values increase the LFO rate.
LFO Trigger (1 & 2)	A negative-going, zero-crossing value resets the LFO to zero.

Modulation Destinations

Destination	Description
Lag Processor (1 & 2)	Inputs for Lag Processors 1 & 2. See page 138.
Lag Rates (1 & 2)	Rate control for Lag Processors 1 & 2. See page 138.
Summing Amp	Input to the Summing Amp. See page 307.
Switch	Input to the Switch processor. See page 307.
Absolute Value	Input to the Absolute Value processor. See page 307.
Diode	Input to the Diode processor. See page 307.
Flip-Flop	Input to the Flip-Flop processor. See page 307.
Quantizer	Input to the Quantizer. See page 307.
Gain 4x	Input to the 4x Amplifier. See page 307.
Key Timer Rate	Input rate control for the Poly Key Timer. See page 86.
Oscillator Speed	Speed percentage modulation for TwistaLoop playback.
FuncGen Rate 1-3	Controls the rate between each step of function generators 1, 2 or 3. See page 143.
FuncGen Retrig 1-3	Restarts the function generator when the value crosses zero in the positive direction.
FuncGen Length 1-3	Modulates the End Step for function generators 1, 2 or 3. See page 143.
FuncGen Dir. 1-3	Changes the forward-backward direction for function generators 1, 2 or 3. See page 143.
Cord 1 - 36 Amount	Controls the amount of PatchCords 1-36.

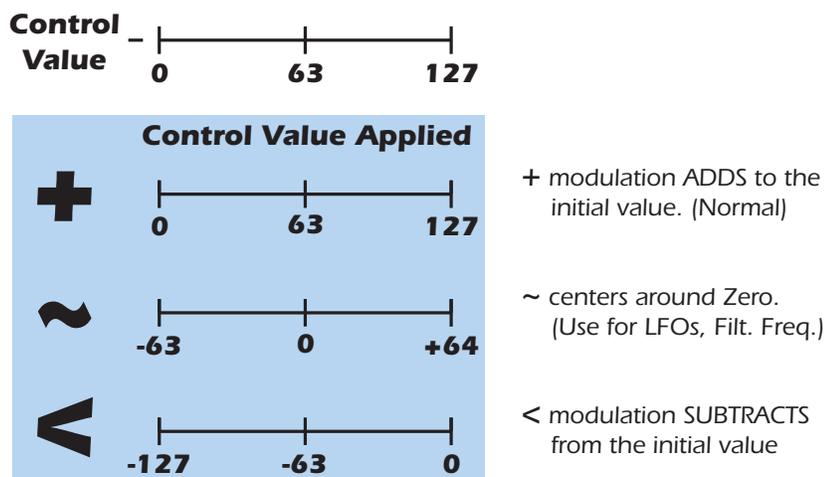
Modulation Source Polarity

On a few of the modulation sources you'll see the following suffixes: +, ~, <. These designate the polarity of the modulation source. A plus (+) means that the source goes from zero to maximum value.

The (~) symbol indicates that the source is bidirectional. In the case of an LFO, the LFO would go both positive and negative, centering on the zero line.

The (<) symbol indicates that the source only subtracts from the initial value of a destination. This type of control is designed to be used for velocity-to-volume control where you want to keep as much headroom as possible.

The following chart shows the three types of polarity.



Summing Nodes

All the modulation inputs on Emulator X are summing nodes. This means that you can connect as many modulation sources as you want to an input (such as Pitch or AmpVol). Modulation sources are simply added algebraically — connecting two knobs one set to -100 and the other set to +100 yields a net value of zero.

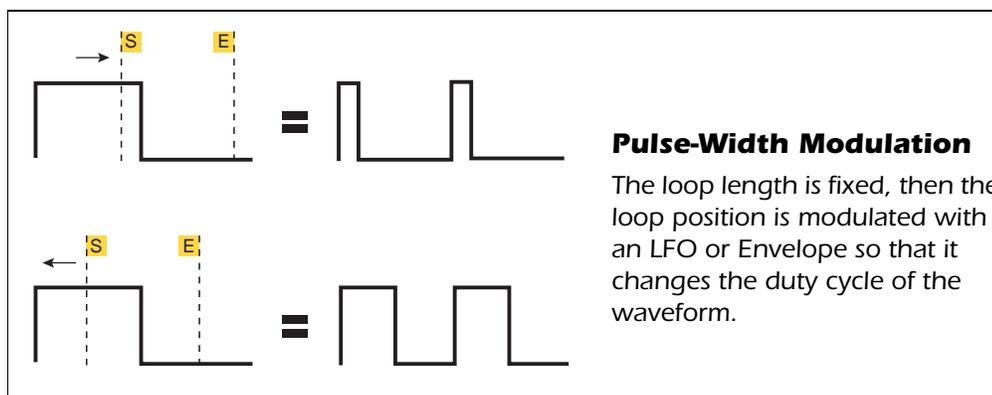
PatchCord Lore

- Real-time events happen first on lower numbered Cords, so it is usually a good idea to put important functions on lower numbered cords and to place components of complex modulations patches in the proper order in the Cords section.
- Negative Cord amounts invert envelopes and LFO waveforms.
- To invert the effect of a controller knob, set its Cord to -100 and add DC with a Cord value of +100.

Sample Loop Modulation

The sample loop position can be modulated using a Cord. [See “Modulation Destinations” on page 159.](#) The loop length remains constant as the loop is modulated. This feature can be used to implement pulse width modulation. Waveforms other than rectangular shapes can be used to implement wavetable scanning synthesis.

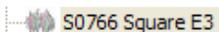
The basic idea of pulse-width modulation using the Emulator X is to start with a looped single-cycle wave, then dynamically move the loop location so that different portions of the waveforms are included. Changing the loop length will change the pitch of the sample, but this is easily corrected using the Transpose and Tuning controls in the Voice.

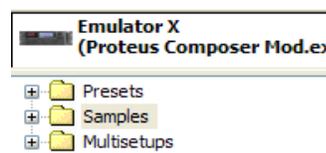


► To Pulse-width Modulate a Square Wave

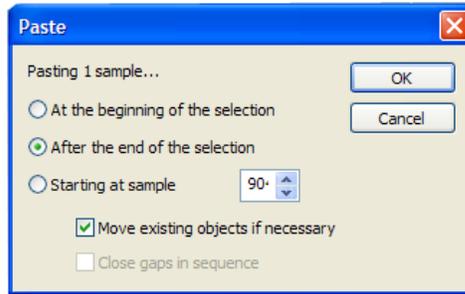
1. Turn **Streaming Off** in the Options, Preferences menu. Pulse width modulation needs to have the entire sample loaded into RAM for proper operation.
2. Load the Proteus X Composer bank.

Copy the Sample

1. Select the **Samples** folder in the Tree.
2. Open the **Samples** folder by clicking on the (+) symbol.
3. Locate sample **S0766 - Square E3** in the Tree, and select it. 

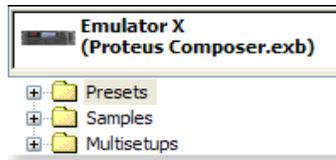


4. Select **Copy** from the Edit menu.
5. Select **Paste** from the Edit menu. The following dialog box appears.
6. Choose **Pasting 1 sample** "After the end of the selection." Also choose "Move existing objects if necessary."
7. A duplicate of Square E3 appears in sample location S0767.
8. **Right-click on S0767**, then choose **Rename (F2)**.
9. Rename the sample to **PWM E3**.



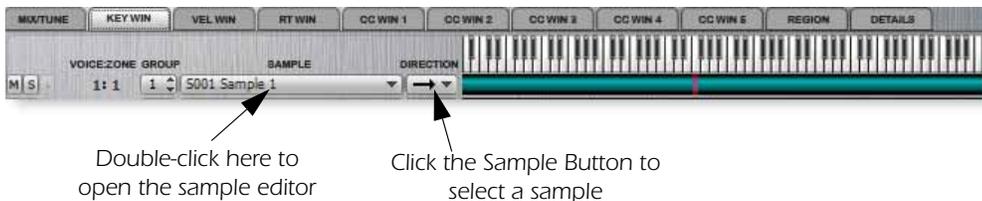
Create a new Preset

10. Select the **Preset** folder in the Tree.
11. Select **Presets, New Preset** from the menu bar.
12. **Open the Preset Folder** by clicking on the (+) symbol and scroll down to the bottom of the preset list. A new empty preset has been created at the bottom of the Tree (location P1026).
13. It's always a good idea to rename your new preset to avoid confusion later. (Right-click on the preset name, then choose **Rename** or press **F2**.)



Assign the Sample

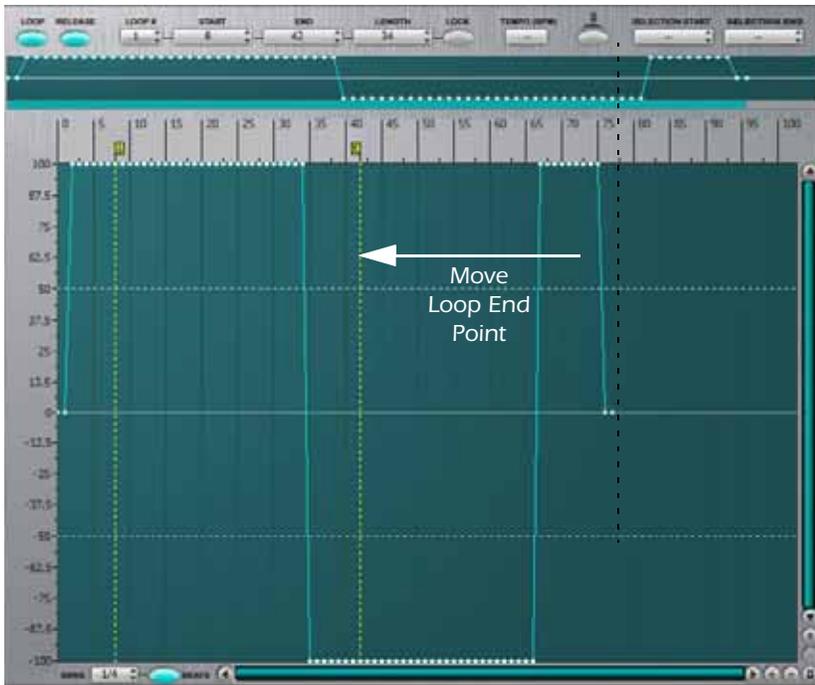
14. Open the preset by clicking on the (+) symbol next to the little [icon] icon.
15. Open the Voices and Zones module. The Key Window appears.



16. Select sample **S0767 - PWM E3** (a pulse width modulation sample).

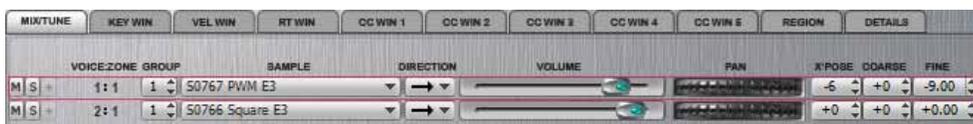
Adjust the Loop Markers

17. **Double-click on the sample name** to open the sample editor.
18. Move the Loop End Point by dragging the marker or by typing in a value into the End field. We have chosen a value of 69 samples for the loop length in the example above. The distance from the start point sets the initial pulse width and pitch of the sample.
19. It's easy to see how the loop will move if you click the **Lock** loop length button and then move the loop start point. Try this and note how the pulse width changes inside the loop markers. When you're finished, move the loop start point back to 8 samples and the end point to 77 samples (Loop Length = 69 samples).



Retune the Sample

20. Unfortunately, because we changed the loop length, the waveform is no longer tuned to E3. Fortunately, this is easy to fix. Click the **Back Arrow**  to jump back to the Key Window.
21. Select **New Voice** (Ctrl+W) from the Preset menu. A new empty voice appears.
22. Click the button next to the voice and select sample **S0766 - Square E3** (the original square wave sample).
23. Click the **Mix/Tune** tab. The Voices and Zones page should look like this:



24. Play the keyboard. Both voices will sound and they will be horribly out of tune. Adjust the **Transpose** and **Fine Tune** controls on **Voice 1** until they are in tune. Now the PWM E3 voice is back in pitch.
25. Select **Voice 2** (S0766 - Square E3) and select **Delete Voices** from the Preset menu since it's no longer needed.

Connect the Modulation Cords

26. Select the **Voice Processing** page from your new preset (P1025) in the Tree. The voice processing page appears.
27. Set up the Cords as shown at left. You only need to change two parameters. Set Cord 3's destination to LFO_ to **Sample Loop**, and change the Amount of Cord 4 to **-2.00**.
28. Play the keyboard and turn up the Mod Wheel. You should now be hearing rich, luscious pulse width modulation.
29. You might want lower the frequency of LFO 1 for a better chorale effect.



Assign Groups

The Assign Groups can be divided into two types:

- **Output Assign Groups** - Set the maximum number of output channels that a voice can use. These groups protect important parts, or cause intentional voice "stealing".
- **Poly Key Assign Groups**- Limits the number of times a voice will sound when the same key is repeatedly pressed. Conserves available polyphony by limiting the number of output channels assigned to a single key.

◆ **Note:** "Output Channels" in this context is defined as channels of polyphony, or the number of sounds that can be simultaneously played.

Output Assign Groups

This function allows you to assign a certain number of output channels to each voice. By assigning all voices in the preset to assign groups, important parts can be protected from being "stolen" by more recently played keys. Or, a voice such as an open high hat can be assigned to a mono channel so it will be cancelled by a closed high hat. Voices will rotate within their assigned bin of channels not interfering with other bins. Poly All is the default mode. The modes are as follows:

Output Assign Groups

Poly All	Notes are played polyphonically with dynamic channel assignment, using all available polyphony.
Poly 16 A-B	Two bins of 16 channels each. Notes are played polyphonically with dynamic channel assignment, using no more than 16 channels.
Poly 8 A-D	Four bins of 8 channels each. Notes are played polyphonically with dynamic channel assignment, but using no more than 8 channels.
Poly 4 A-D	Four bins of 4 channels each. Notes are played polyphonically with dynamic channel assignment, but using no more than 4 channels.
Poly 2 A-D	Four bins of 2 channels each. Notes are played polyphonically with dynamic channel assignment, but using no more than 2 channels.
Mono A-I	Nine monophonic channels. Any voices assigned to the same letter interrupt each other, but do not affect other voices.

Poly Key Assign Groups

The Poly Key Assign groups limit the number of times a voice will be duplicated when the same key is repeatedly pressed. The Poly Key Groups limit polyphony per key and allow you to make better use of the available polyphony of the instrument.

When using the default setting of “Poly All”, repeatedly playing a single key can eat up large numbers of voices, especially with pianos, cymbals, snares and other sounds where the envelope release parameter is used. The Poly Key Assign Groups allow you to impose a limit (1, 2, 3, 4, 5, 6 or 8) on the number of simultaneous notes that a single key will trigger.

The polyphony count is noted in the Poly Key item (8, 6, 5, 4, 3, 2, 1)—Poly Key 8 limits the voice count to 8 voices per key (16 for stereo), Poly Key 4 limits the voice count to 4 voices per key (8 for stereo), Poly Key 2 limits the voice count to 2 voices per key (4 for stereo).

In addition, there are four different “bins” for each voice group labelled A, B, C, D. These four bins keep completely separate voice counts and allow complex control of polyphony. For example, if 3 voices are assigned to a single key, 1 voice could be assigned to a 1-voice group and the other 2 voices could be assigned to a 8-voice group. The 1-voice will retrigger itself on each key press, while the 2 voices assigned to the 8-voice group will be allowed to ring out, producing a chorus effect.

As another example, suppose you had 4 voices assigned to a key and wanted to limit the polyphony of the key to 4 notes ($4 \times 4 = 16$ voices max). By assigning each of the four voices to a different 4-voice bin, each voice on the key is allowed to play only four simultaneous notes.

Poly Key Assign Groups

Poly Key 8 (A-D)	Poly Key 8 limits the voice count to 8 voices per key (16 for stereo)
Poly Key 6 (A-D)	Poly Key 6 limits the voice count to 6 voices per key (12 for stereo)
Poly Key 5 (A-D)	Poly Key 5 limits the voice count to 5 voices per key (10 for stereo)
Poly Key 4 (A-D)	Poly Key 4 limits the voice count to 4 voices per key (8 for stereo)
Poly Key 3 (A-D)	Poly Key 3 limits the voice count to 3 voices per key (6 for stereo)
Poly Key 2 (A-D)	Poly Key 2 limits the voice count to 2 voices per key (4 for stereo)
Poly Key 1 (A-D)	Poly Key 1 limits the voice count to 1 voice per key (2 for stereo)

► Conserving Polyphony Using the Poly Key Groups:

The Poly Key Group feature is designed for building more efficient presets from the ground up, but if you have an existing preset which seems to be consuming too much polyphony, you can try this quick fix which will work in many (but not all) cases:

1. Select the preset and **Duplicate** it.
2. Using the Copy, go to Voices and Zones and **Select All**.
3. Go to Voice Processing and set the Assign Group to **Poly Key 2A** (or Poly Key 1A).
4. Play the preset—if it sounds too dry— change the setting to **Poly Key 3A**.

5. Go to the original preset and **play C3** over and over while watching the voice display at the bottom of the Emulator X screen.
6. Do the same with your new preset to see the polyphony savings and make sure that you haven't altered the sound of the preset in a negative way. Experiment!
7. Save the Bank.

Keyboard Modes

The Keyboard mode controls how the voice will be triggered from the keyboard. Poly Normal mode is the default mode which allows you to play chords.

The eight different solo modes provide the playing action of a monophonic instrument—only one note can be played at a time. Monophonic wind, brass and string instruments often sound more realistic with a single note playing. Synthesizer lead lines also require solo mode. There are solo modes provided which can be used with monophonic synthesizer sounds or to create other performance effects.

The Release modes trigger a voice when the key is released. Many acoustic instruments, such as a harpsichord, generate sounds when a key is released. To use this feature, create a special "Release Voice" and turn one of these modes on for that voice.

▼ Solo modes sometimes require extra Pre-roll [\(page 22\)](#) length to perform properly. If you are having problems with the solo modes, try increasing the Pre-roll.

Poly Normal	Standard polyphonic mode. Any number of notes can be played.
Multiple Trigger	Last note priority. No key-up action. Retrigger envelopes and samples when a key is pressed.
Melody (last)	Last note priority. No key-up action. First solo note: Envelopes start at Attack segment from zero. Samples start at the beginning. If previous note is releasing: Envelopes start at Attack segment, but from current level. Samples start at the beginning. When playing "Legato": Envelopes continue from current segment and level. Samples start at the loop or the beginning if unlooped.
Melody (low)	Same as Melody (last), but with low-note priority. Newly played keys which are higher than the lowest solo key held will not sound.
Melody (high)	Same as Melody (last), but with high-note priority. Newly played keys which are lower than the highest solo key held will not sound.
Synth (last)	Similar to Melody (last) with key-up action. When currently sounding solo key is released while still holding other keys, the highest held solo key sounds in a Legato fashion (envelopes not retriggered).
Synth (low)	Same as Synth (last), but with low note priority. When the currently sounding solo key is released when other keys are still held, the lowest held solo key will sound in a Legato fashion. (MiniMoog).
Synth (high)	Same as Synth (last), but with high-note priority. When the currently sounding solo key is released when other keys are still held, the highest held solo key will sound in a Legato fashion.
Fingered Glide	Same as Synth (last), except that Glide is disabled when playing Staccato, enabled when playing Legato.

▼ You must let the note decay completely when changing envelope generator times in solo mode to hear the changes.

Poly Release Trig Release Vel	Polyphonic action, but the voice triggers only when the key is released. The key velocity is determined by how fast the key is released.
Poly Release Trig Note-on Vel	Polyphonic action, but the voice triggers only when the key is released. The key velocity is determined by how fast the key is pressed
Solo Release Trig Release Vel	Monophonic action, but the voice triggers only when the key is released. The key velocity is determined by how fast the key is released
Solo Release Trig Note-on Vel	Monophonic action, but the voice triggers only when the key is released. The key velocity is determined by how fast the key is pressed

Latch Mode

When Latch mode is On, pressing a key once will latch the key on. Pressing the key again will unlatch the key. This mode was originally designed for doing sound effects work, but you can probably find many other uses for it, such as latching sampled grooves and beats. Any voices assigned to the same key as a latched voice will also latch.

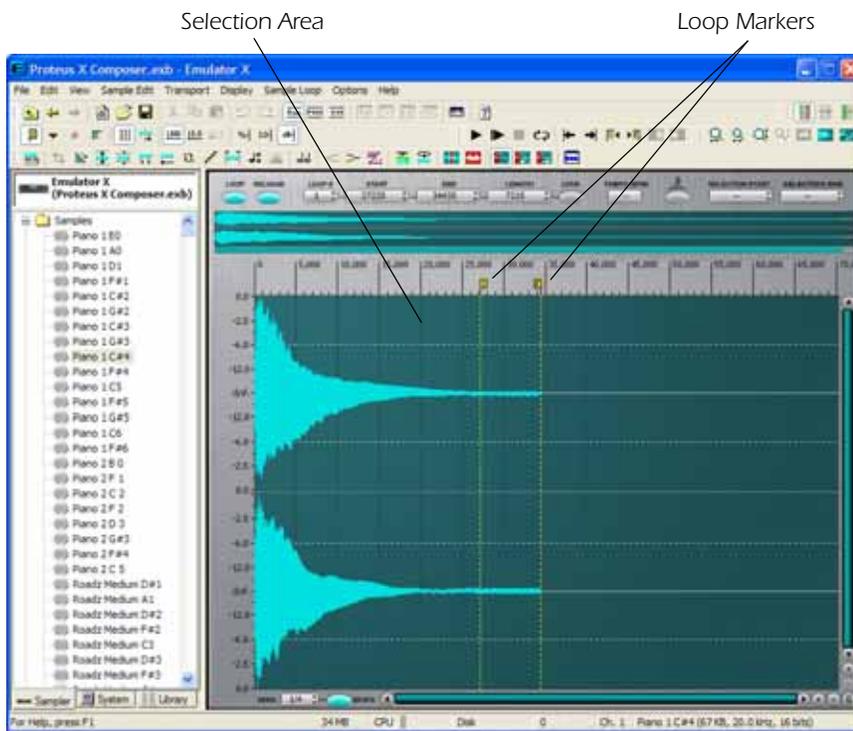
▼ **Caution:** Using Latch Modes together with the Solo Modes may cause unpredictable results.

7 - Editing a Sample

Sample Editor Window

The Sample Editor allows you to modify the actual sample data in order to perform functions such as: sample looping, sample truncation and splicing, equalization, time compression, sample rate conversion and many others. In the Sample Editor module we are working with the most basic element of the Emulator hierarchy, the sample. It is important to note that every one of these functions except basic looping modifies the sample data.

Even though you can undo these operations, it's a good idea to make a copy of the sample before modifying the original if you want to keep an original version of the sample.



► To Get to the Sample Editor Window:

- Click on any sample in the tree to view that sample in the Sample Editor window. A stereo sample is shown above.
- Double-click on any sample in the "Voices and Sample Zones" window.
- You can also select samples by double-clicking on the sample name in the Voice Processing window, or by selecting a sample and choosing "Open Sample" from the Preset window.

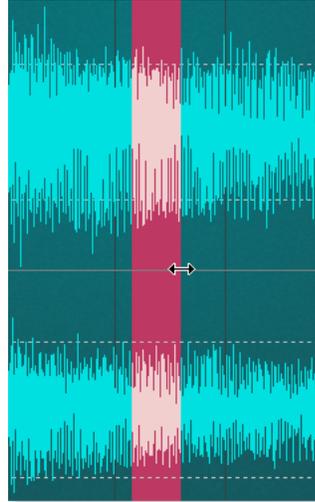
◆ **Note:** 24-bit samples are shown as 32-bit because they are internally stored and processed as 32-bit numbers.

► To Select an Area of a Sample for Editing (Mono):

1. Click anywhere on the sample display to place the cursor.
2. Click and drag the cursor to select an area of the sample for processing.
3. Or... press and hold the shift key after placing the cursor, then click again on the waveform to select the area between the two mouse clicks.

► **To Select Both Sides of a Stereo Sample**

1. Place the cursor near the center of the display, as shown at right.
2. **Click and drag** the cursor to select an area of the sample for processing.
3. Or... **press and hold the shift key** after placing the cursor, then click again on the waveform to select the area between the two mouse clicks.



► **To Select the Left Side of a Stereo Sample**

1. Place the cursor in the upper 1/3 of the display, as shown at right. The letter L appears next to the cursor.
2. **Click and drag** the cursor to select an area of the sample for processing.
3. Or... **press and hold the shift key** after placing the cursor, then click again on the waveform to select the area between the two mouse clicks.



► **To Select the Right Side of a Stereo Sample**

1. Place the cursor in the lower 1/3 of the display, as shown at right. The letter R appears next to the cursor.
2. **Click and drag** the cursor to select an area of the sample for processing.
3. Or... **press and hold the shift key** after placing the cursor, then click again on the waveform to select the area between the two mouse clicks.



Exporting & Importing Samples

The Export function allows samples in the bank to be exported as Wave or AIFF files. Samples can be imported using the import option in the File menu. You can also simply drop a WAVE, MP3, REX 2 or AIFF file over the Emulator X icon, onto the Multisetup screen or onto the Voices and Zones screen.

► To Export a Sample from the Bank

- Right-click any sample in the Tree or List views.
- Select a sample, then select Export from the Sample Edit menu.

► To Import a Sample into the Bank

- Select Import Samples from the File menu. Audio files (Wave, MP3, AIFF, REX 2) can be auditioned from the import browser.
- Drop a Wave, AIFF, MP3, or REX 2 file over the Emulator X icon.
- Drop a Wave, AIFF, MP3, or REX 2 anywhere in the Voices and Zones screen. This creates a new voice with that sample with an original key at C3 and a Key Window range from C-2 to G8.
- Drop a Wave, AIFF, MP3, or REX 2 over the keyboard at the bottom of the Voices and Zones screen. This creates a new voice with the sample mapped (including original key) to the note where you drop it.
- Drop a Wave, AIFF, MP3, or REX 2 file over the Multisetup screen, or the Multisetup button (above the tree), bringing up the Import Samples dialog box.
- Drop a Wave, AIFF, MP3, or REX 2 file over the Samples folder in the tree. The sample imports at its original sample rate.

▼ **Vista Users:** To Audition samples, you may need to download and install the Vista Codec Package 4.5.4.

■ To import other sample formats, see [“Translator Emulator X Converter” on page 319.](#)

▼ **64-bit Users:** To Import REX files, you must use the 32-bit version of Emulator X3.

Exporting a Tempo Map

The Tempo Map of a “beat analyzed” sample ([page 209](#)) can be exported as a standard MIDI file (.mid). This file can then be imported along with the corresponding audio file to your sequencer in order to create a tempo that’s perfectly aligned to the audio file you analyzed.

Once imported, your sequencer’s bars and beats (and click track) will be perfectly aligned to the tempo of your audio file. This means your sequencer “grooves” with you (as opposed to you trying to groove to your sequencer’s rigid click track).

So give it a shot— analyze a drum beat, or a guitar track you recorded in Emulator X, run beat analysis and create a tempo map. Once you import the audio and MIDI info into your sequencer, your sequencer will play right along.

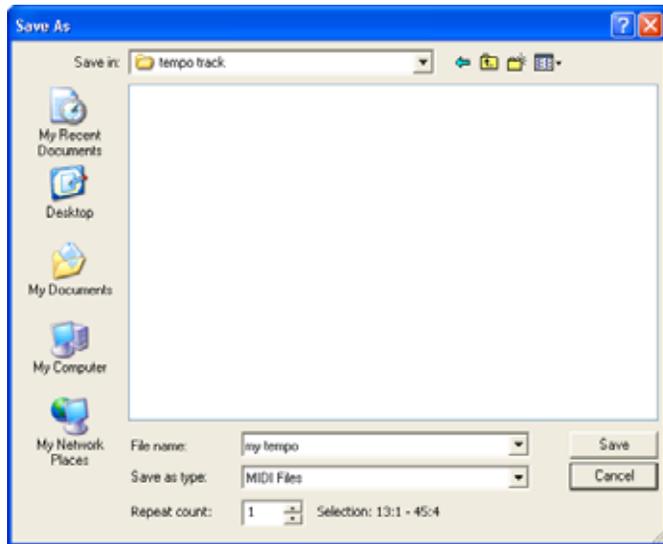
When you export a tempo map, the tempo map updates the tempo information at each Beat Marker. By setting the Tempo Granularity ([page 214](#)) of the Beat Analysis, you can adjust how closely the Tempo Map follows the source material.

► To Export a Tempo Map

1. Select a sample in the Sample Editor.
2. If you haven’t already, run **Beat Analysis** on the sample. See [“Running Beat Analysis” on page 209](#) for more information.

7 - Editing a Sample Exporting a Tempo Map

3. In the sample edit window, select the beats you want included in your tempo map file. You should make sure your selection begins and ends on a beat marker.  If your sample has an area at the beginning with no assigned tempo, DO NOT select it.
4. Select **Export Tempo Map...** from the **Sample Edit** menu. A popup dialog box appears:



5. In the dialog box, select the number of times you want your tempo track to repeat.
6. Select a location for your Tempo Map and click **Save**.

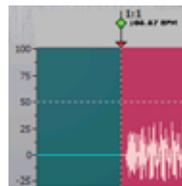
Tips for Exporting Tempo Maps

If you've run beat analysis on an audio file, and exported a tempo track, you should be able to import the tempo track and the corresponding audio file to your favorite sequencer.

However, if the length of your audio file differs from the length of your exported tempo in any way, you'll need to create a new audio file to import into your sequencer.

You must create a corresponding audio file if:

- Your exported tempo doesn't cover the entire audio file. For example, you exported only a few measures of tempo data. You would then need to export the audio for the same few measures.
- The first beat begins after the beginning of the sample. This can happen if your sample has some silence at the beginning, or if beat analysis didn't correctly analyze the sample, as illustrated to the right.



► To Create a Corresponding Audio File

1. Copy and paste the exact same portion of the sample you selected for exporting a tempo map. See ["To Copy a Section" on page 181](#).
2. Paste the selected area as a new sample. See ["To Paste as a New Sample" on page 183](#).
3. Export the audio using the **Export...** function in the **Sample Edit** menu. See ["To Export a Sample from the Bank" on page 171](#).

Importing a Tempo Track to a Sequencer

Most commercial sequencing applications should support MIDI tempo tracks. Here's how to do it with Cubase or Sonar.

► To Import a Tempo to Steinberg Cubase

1. Launch Cubase and create a new project.
2. Import the audio file that corresponds with the tempo track you created. From the **File** menu, select **Import, Audio File...** Make sure the audio file is aligned with the very beginning of the project (bar 1, beat1).
3. Import the MIDI tempo track you created. From the **File** menu, select **Import, MIDI File...**
While it may appear that nothing happened, in fact, you just created a tempo track. You'll notice your timeline is now aligned with your imported audio file, and if you play back audio with the metronome on, it should be lined up perfectly.
4. You can view the tempo track by clicking the **Project** menu, and selecting **Tempo Track**.

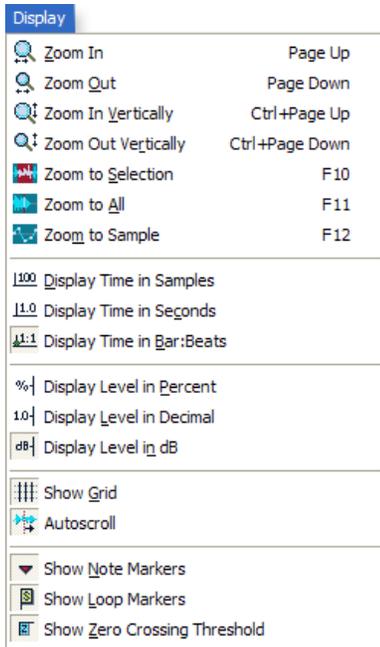
► To Import a Tempo to Cakewalk Sonar

1. Launch Sonar.
2. If you see the Quickstart menu, click **Open Project**. Otherwise, select **Open...** from the **File** menu.
3. Open the .mid file you created as a tempo track. While it may appear that you created a blank new project, in fact, you just created a new project with a tempo track that uses the tempo information from the .mid file.
4. Import the audio file that corresponds with the tempo track you created. From the **File** menu, select **Import, Audio...** Make sure the audio file is aligned with the very beginning of the project (bar 1, beat1).
Your timeline should be aligned with your imported audio file, and if you play back audio with the metronome on, it should be lined up perfectly.
5. You can view the tempo track by clicking the **View** menu, and selecting **Tempo**.

Important: If you want to save the Sonar project, you should choose **Save As...** rather than **Save**. From the **Save as type** drop-down menu, choose **Normal**. Otherwise, you'll overwrite your .mid file.

Display Menu Items

These options are available in the Display menu. If you Right-click in the vertical or horizontal units area of the waveform display a popup menu appears with the same options.



Zoom In (Page Up)

Magnifies the horizontal scaling to show more detail in the waveform. You can also use the Zoom controls in the lower right corner of the waveform display to increase or decrease horizontal resolution.

■ Tip: You can also use the mouse wheel to zoom in and out.

Zoom Out (Page Down)

Reduces the horizontal scaling to show less detail but more of the entire waveform.

Zoom In (Ctrl +Page Up)

Magnifies the displayed height of the waveform to show more detail.

Zoom Out (Ctrl +Page Down)

Reduces the displayed height of the waveform.

Zoom To Selection (F10)

Automatically scales the display so that the selected area is visible.

Zoom To All (F11)

Automatically scales the display so that the entire waveform is visible.

Zoom To Sample (F12)

Automatically scales the display to the highest magnification.

■ Tip: You can also zoom to all by clicking the  button in the bottom-right corner between the zoom sliders.

100 Display Time in Samples

Units are expressed in samples beginning from the start of the sample.

1.0 Display Time in Seconds

Units are expressed in seconds beginning from the start of the sample.

1:1 Display Time in Bar:Beats

Displays Bar and Beat Markers instead of samples or seconds.

% Display Level in Percent

Units are expressed in percentages with 100% equal to full code and 0% as the baseline.

1.0 Display Level in Decimal

Units are expressed in decimal form with 1.0 equal to full code and 0 as the baseline.

dB Display Level in dB

Units are expressed in decibels (headroom) with 0.0 equal to full code and infinite as the baseline.

 **Show Grid**

Show or hide the vertical black grid lines across the waveform display.

 **Autoscroll**

Autoscroll, when on, keeps the cursor line in the center of the waveform display.

 **Show Note Markers**

Show or hide Note Onset Markers.

 **Show Loop Markers**

Show or hide Loop Markers.

 **Show Zero Crossing Threshold**

Shows or hides the Zero-crossing Threshold markers located near the zero vertical line on the waveform display. The zero-crossing threshold can be adjusted to accept sample points which are close to zero. See [page 206](#) for additional information.

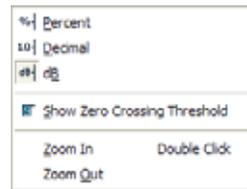
■ **Tip:** You can hold the Ctrl key while scrolling the mouse wheel to scroll right and left.

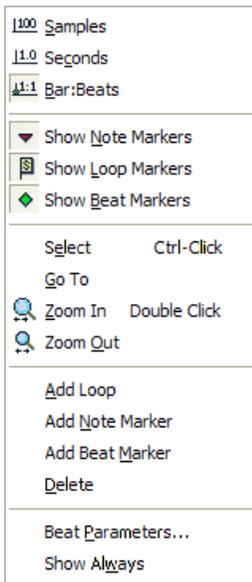
Vertical Units

If you right-click in the vertical units area of the waveform display the popup menu shown at left appears.

Zoom In Increases the Vertical magnification by 2
Double Click times with each selection.

Zoom Out Decreases the Vertical magnification by 2
 times with each selection.



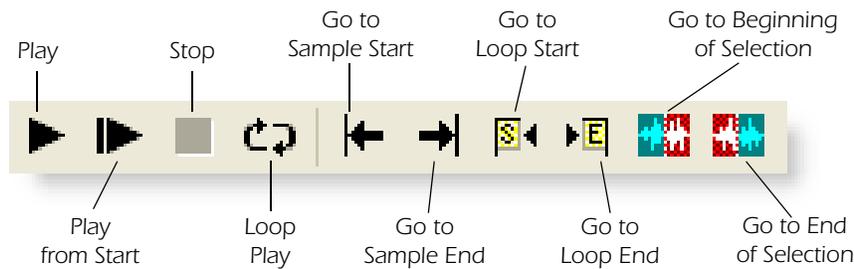


Horizontal Units

If you right-click in the horizontal units area of the waveform display the popup menu shown at left appears.

- ◆ Show Beat Markers** Show or hide Beat Markers.
- Select** Ctrl-Click Right-click on a Beat Marker, Note Onset, or Loop Marker to select.
- Go To** Moves the cursor to the selected region.
-  **Zoom In** Increases the horizontal magnification by 2 times with each selection.
Double Click
-  **Zoom Out** Decreases the horizontal magnification by 2 times with each selection.
- Add Loop** Adds a pair of Loop Markers that you can place wherever you want.
- Add Note Marker** Adds a Note Onset Marker at the current cursor position. See [page 212](#).
- Add Beat Marker** Adds a Beat Marker at the current cursor position. See [page 212](#).
- Delete** (Note/Beat Marker) Deletes the selected note or beat marker.
- Beat Parameters** This function allows you to change the beat number, beats-per-bar or tempo granularity. (Right-click on a Beat Marker to enable.)
- Show Always** Shows the selected marker regardless of view settings. Right-click a marker to enable.

Transport/Loop Controls



Play (from Cursor)

The sample plays from the cursor location and continues to play between the loop markers. This button latches on until the Stop button or Space bar is pressed. Press and hold the space bar to play the sample from the computer keyboard.

Play from Start

The sound plays from the start of the sample regardless of the cursor position.

Stop

Stops playback immediately. Press the Space bar to stop the sample from the keyboard.

Loop Play

Plays the sample repeatedly between the cursor line and the end of the sample when the Play button is pressed. See [“How to Loop a Sample” on page 207](#).

Plays the sample repeatedly between the start and the end of the sample when the Play from Start button is pressed.

Go to Sample Start

Positions the cursor at the start of the sample.

Go to Sample End

Positions the cursor at the end of the sample.

Go to Loop Start

Positions the cursor at the loop start point.

Go to Loop End

Positions the cursor at the loop end position.

Move Start to Previous Zero	Left Ctrl+ <-
Move Start to Next Zero	Left Ctrl+ ->
Move End to Previous Zero	Right Ctrl+ <-
Move End to Next Zero	Right Ctrl+ ->
Autocorrelate Loop	Ctrl+K

Move Start to Previous Zero

Moves the Loop Start Marker toward the beginning of the sample to the next zero-crossing point. (Left Ctrl + <-)

Move Start to Next Zero

Moves the Loop Start Marker toward the end of the sample to the next zero-crossing point. (Left Ctrl + ->)

Move End to Previous Zero

Moves the Loop End Marker toward the beginning of the sample to the next zero-crossing point. (Right Ctrl + <-)

Move End to Previous/Next Zero

Moves the Loop End Marker toward the end of the sample to the next zero-crossing point. (Right Ctrl + ->)

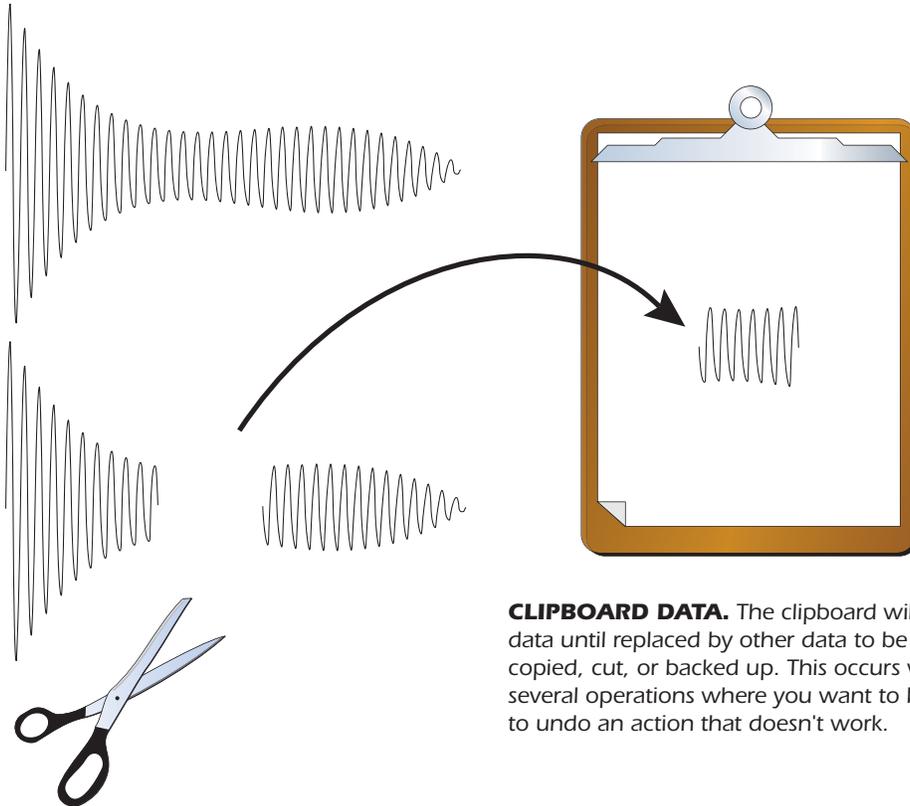
Autocorrelate Loop

The computer analyzes the waveform around the loop points you have specified and moves the start point to find a section of the wave that closely matches the section around the end point. Auto correlation can be use again and again, with the computer moving the analysis window slightly each time to zero in on the perfect loop.

For more information, see ["How to Loop a Sample" on page 207](#).

Cut, Copy, Paste & Delete

Imagine a printout of a sample. Now imagine that you have a pair of scissors capable of cutting a piece out of that sample (or a copy of that piece if you don't want to affect the original), as well as a clipboard to neatly hold the cut or copied piece. Now imagine that you can either perfectly insert the clipboard contents at any given point in any sample, or mix the clipboard contents in with any sample, starting at any designated point. The Emulator X can do all this, and all electronically. Auto Correlation and Crossfade functions are also provided for seamless transitions between the cut, copied, and pasted parts.



CLIPBOARD DATA. The clipboard will retain data until replaced by other data to be copied, cut, or backed up. This occurs with several operations where you want to be able to undo an action that doesn't work.

Undo and Redo

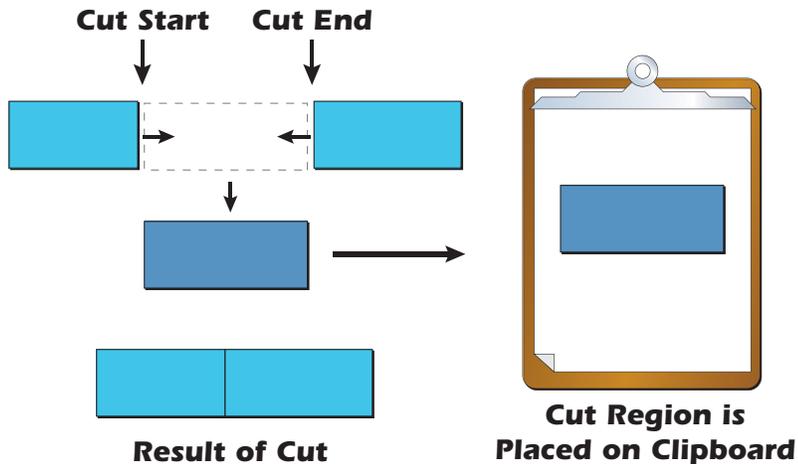
During all sample edit operations, the sample being processed will automatically be backed up and stored on the hard disk. If you do not like the results of the processing, you can call up the Undo function, and restore the original sample. (Ah, if only life itself were so simple.) Best of all, the processed sample will now be backed up. Thus, if you decide you liked the processed version better after all, you can actually undo the undo. (Redo.) There are 100 levels of Undo.

Typical Applications

Typical cut/copy/paste applications would be to splice the beginning of one sample to the end of another, or to mix two samples together to conserve memory. (*Do this by copying an entire sample, then pasting it at the beginning of the second sample using the mix option.*) You can splice an attack transient on to a synthesizer waveform loop. Another possibility is, flanging and chorusing. Paste (mix) a sample to itself, offset from the beginning by a few hundred samples or so to thicken up the sound.

Cut Section

The Cut function removes a section of a sample, and stores the cut portion in a special part of memory called the clipboard. Portions of a sample can be cut, copied, and pasted to other samples or placed into a new sample. Cut can also be used to simply remove a section of the waveform.



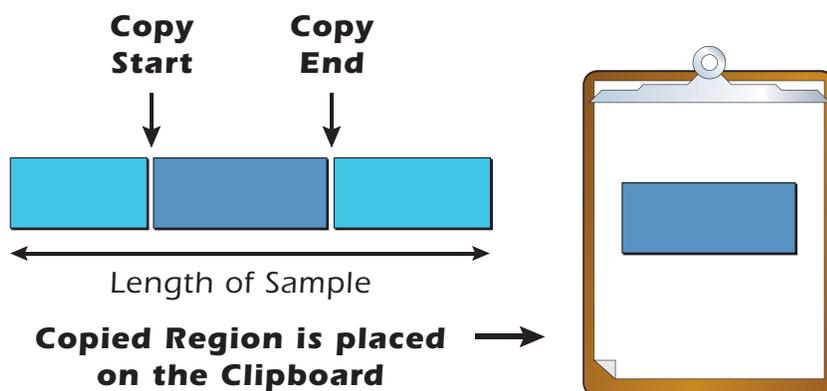
► To Cut a Section

1. Click once on a sample icon in the Tree to select it for editing. The waveform display appears.
2. Magnify the waveform as desired using the Zoom button below the display. Press the "Go To Start" or "Go To End" buttons to quickly jump to the beginning or the end of the sample.
3. Select the section of the sample you wish to cut. [See page 169.](#)
4. Select Cut from the Edit menu or Toolbar. The selected section will be removed from the sample and placed onto the clipboard.
5. Use the Undo function (Ctrl+Z) in the Edit menu to if you don't like the results of the cut.

■ Press the Play button to hear the sample as you adjust the selection area.

Copy Section

The Copy function allows you to duplicate a section of a sample, and store that duplicate in a special part of memory called the clipboard.



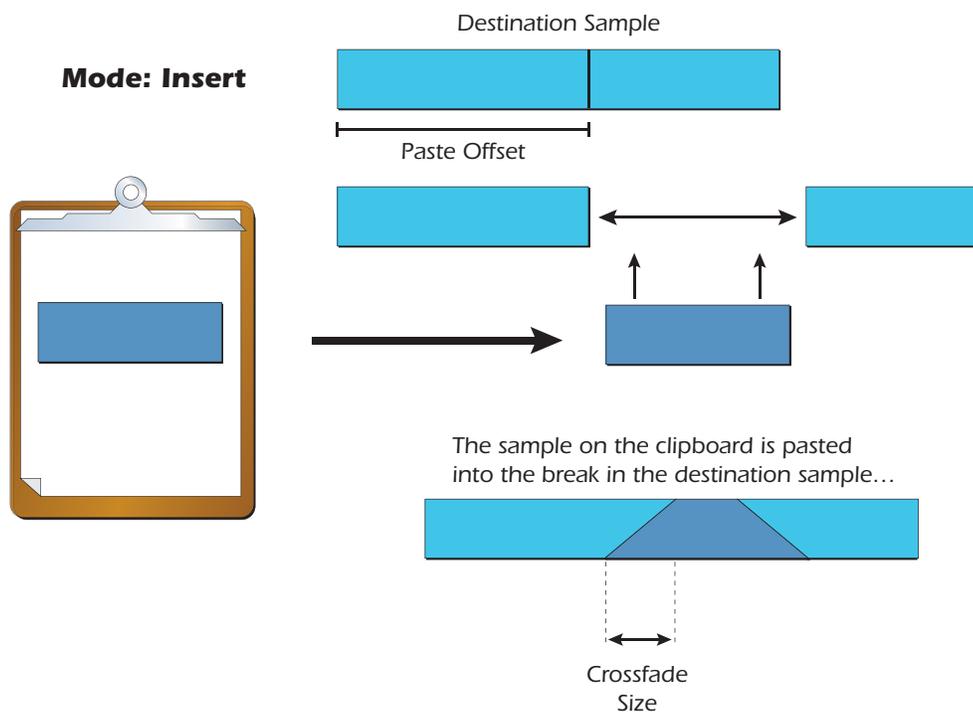
► To Copy a Section

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. **Magnify** the waveform using the Zoom button below the display. Press the “Go To Start” or “Go To End” buttons to quickly jump to the beginning or the end of the sample.
3. Select the section of the sample you wish to copy. [See page 169.](#)
4. Select **Copy** from the Edit menu or Toolbar. The selected section will be duplicated and placed onto the clipboard.
5. Use the **Undo** function (Ctrl+Z) in the Edit menu to undo the copy process.

- Press the Play button to hear the sample as you adjust the selection area.

Paste Section

Paste takes the clipboard contents and either inserts them in a sample at a specified point, or mixes them with a sample starting at a specified point.



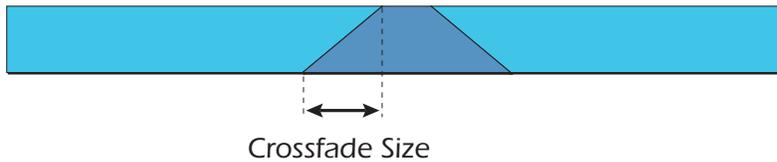
...or crossfaded in and out of the break in the destination sample.

► To Paste a Section

1. **Magnify** the waveform as needed using the Zoom function.
2. Place the cursor at the point where you want to paste the clipboard contents.
3. Select **Paste** from the Edit menu or Toolbar. The following dialog box appears.

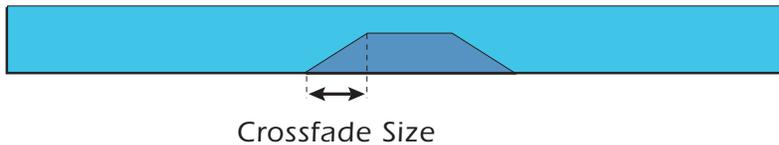
- Select Zero Crossings in the waveform by using the Control keys. See [page 206](#) for details.

Mode: Crossfade Mix **Source Amount:** 100%



The clipboard contents will be gradually mixed into the destination sample and will replace the destination when the fade reaches 100%.

Mode: Crossfade Mix **Source Amount:** 50%



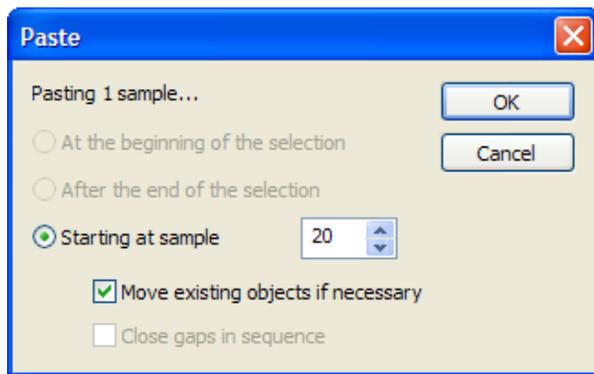
The clipboard contents will be gradually mixed into the destination sample and reach a maximum of 50-50 mix.

6. Select **OK** to perform the Paste or **Cancel** to cancel the operation.
7. Use the **Undo** function (Ctrl+Z) in the Edit menu if you're not happy with the results of the paste operation.

► **To Paste as a New Sample**

Suppose you wanted to copy a section of one sample and create a new sample from the clipboard contents. Here's how.

1. **Copy** the desired section to the clipboard (Ctrl+C).
2. Select the **Samples** folder from the Tree.
3. Choose **Paste** from the Edit menu (Ctrl+V). The dialog box show below appears.



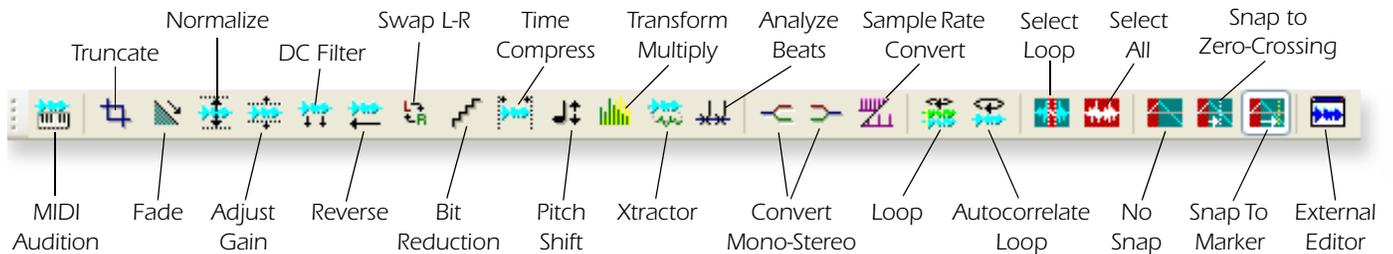
4. **Choose the number in the list** where you want the sample to be placed. If you choose an existing sample number and "Move existing objects if necessary" is selected, the existing samples in the back will be renumbered to accommodate your new sample.
5. **Click OK.**

More Information about Paste

- You can cut, copy and paste multiple samples from the List View section of the screen.
- Pasting at the insertion point moves the sample after the insertion point and **Inserts** the clipboard contents.
- Pasting into a selected region **Overwrites** for the length of the clipboard contents.
- If you paste into a selected region which is shorter than the clipboard contents, only the selected region will be overwritten. The remainder of the clipboard sample will be **Inserted** after the selected region. (Data will not be modified out of the selected region.)
- Pasting a mono sample into one side of a stereo sample adds silence to the other side in order to preserve phase linearity.

Sample Edit Tools

The sample edit toolbar displays the most commonly used sample editing tools. These tools can also be accessed from the Sample menu under “Edit”. Select an area of the sample for processing using the mouse, then select the process you wish to perform.



MIDI Audition

MIDI Audition allows you to (monophonically) play the current sample using your MIDI keyboard. The sample can be transposed up or down from C4 which plays the **original pitch**. The transpose feature is useful when looping since sometimes a transposed sample will reveal a tick or pop which was masked at original pitch.

A looped sample will continue to play inside the loop from the point where you released the key. An un-looped sample will play from the beginning as long as you play the entire sample. If you release the key before the end of the sample, the sample will begin at the release point the next time you press a key. This feature is useful to cue up extremely long samples.

- To play the entire sample from the beginning, select the entire waveform (**double-click** on the wave or use **Select All**), then play the keyboard.
- The sample will always play from the beginning of any selected region.
- To play from the beginning of a sample with no region selected, press the **Ctrl** key on the computer keyboard while you play your MIDI keyboard.
- Footswitch 1 (set in Preferences, Controllers) operates as a sustain pedal when MIDI audition is enabled.

► To MIDI Audition a Sample

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. Turn on **MIDI Audition** by pressing the button on the toolbar or by selecting **Audition from MIDI** from the Sample Edit menu.
3. **Play** a key on your MIDI keyboard. The selected sample plays.
4. Play another key. Note that the sample begins playing from the point you released the key.
5. Press the **Ctrl** key while you play the MIDI keyboard to play from the beginning of the sample.
6. Select a region (or the entire sample) to audition only the selected region.

▼ Important:

IntelliEdit **MUST** be turned ON in the MIDI Preferences menu for MIDI Audition to work.

Truncate

Truncation shortens a sample by **keeping the selected area of the sample** and trimming off the beginning and/or end. Truncation is most often used to remove unneeded portions of a sample, but it can also be used to change instrument characteristics such as removing the attack from a plucked string note or isolating a particular section.

- Truncate any silence off the beginning of your samples to avoid slow key response.

► To Truncate a Sample

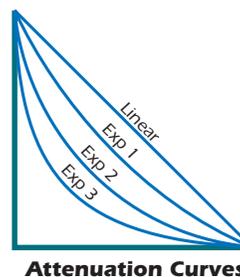
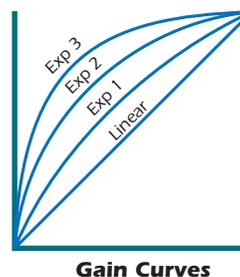
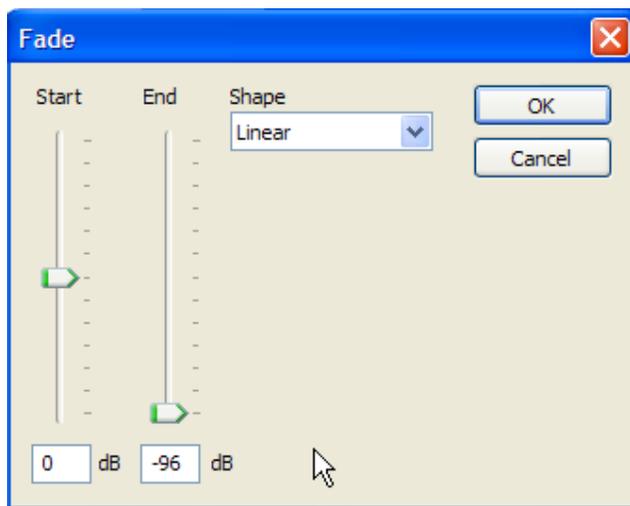
1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. **Magnify** the waveform using the Zoom button below the display. Press the “Go To Start” or “Go To End” buttons to quickly jump to the beginning or the end of the sample.
3. **Select the area** of the waveform you want to keep.
4. Press the **Truncate** button. Everything but the selected area will be discarded.
5. Use the **Undo** function (Ctrl+Z) in the Edit menu if you are not happy with the results.

Fade

Fade allows you to change the gain in a smooth curve, lets you create an artificial decay on percussion samples where the original decay is absent, create an artificial fast attack (such as bowed violin with a slow attack), or clean up background noise when editing dialog.

► To Fade a Sample

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. **Magnify** the waveform using the Zoom button below the display if necessary.
3. **Select the area** of the waveform you want to fade. Press the Play button to hear the sample as you adjust the selection area.
4. Press the **Fade** button. The following dialog box appears.



5. The volume will fade between the selected (start & end) gain/attenuation amounts following the selected curve shape. **Set the start and end amounts and curve.**
6. If the sample is stereo you have the option to fade the right side only, left side only or both.
7. Select **OK** to perform the fade or **Cancel** to cancel the operation.
8. Use the **Undo** function (Ctrl+Z) in the Edit menu if you are not happy with the results.

Normalize

Normalize checks for the loudest peak in the selected section of the sample and raises the overall level of the section so that the loudest peak is at full level (0dB).

Stereo samples are normalized so that the gain is increased equally for both channels until one channel's highest peak reaches 100%. If you wish to normalize each side of a stereo sample separately, use the normalize option in the Adjust Gain tool.

► To Normalize a Sample

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. Usually, you will normalize the entire sample. In this case you don't have to select a specific area.

Selecting a specific region of the sample to normalize is not recommended as this can create a sudden thump or pop at the selection point from the sudden gain increase.

3. Press the **Normalize** button. The amplitude of the sample will be increased to the optimum level.
4. Use the **Undo** function (Ctrl+Z) in the Edit menu if you are not happy with the results.

Adjust Gain

Use this function to raise or lower the level of the selected section of the sample from -96 dB to +96 dB. There is also an option to normalize the section.

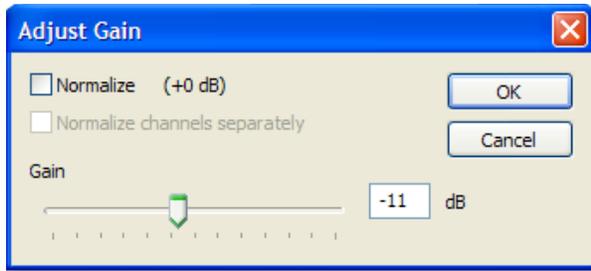
With a stereo file, you can choose to normalize each channel separately. This method boosts the volume differently on each channel without preserving the volume balance between the left and right channels.

► To Adjust the Gain of a Sample

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. Usually, you will adjust the gain of the entire sample. In this case you don't have to select a specific area.

Selecting a specific region of the sample to gain change can create a sudden thump or pop at the selection point caused by the sudden gain increase or decrease. Use the Fade tool to gradually change the gain of a sample section.

3. Press the **Adjust Gain** button. The following dialog box appears.



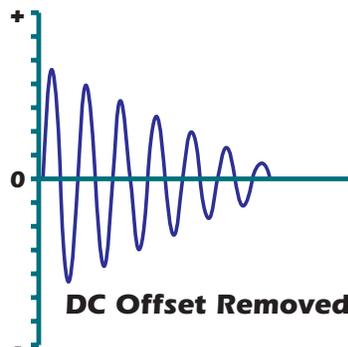
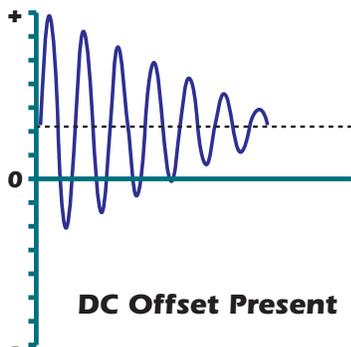
4. Select the **amount of gain** increase or decrease for the sample. Moving the slider all the way left (-inf) silences the sample.
5. If the sample is stereo you have the option to change the gain of the right side only, left side only or both. If you wish to **Normalize** the sample, check the normalize box. If the sample is stereo you have to option to normalize each channel separately.
6. Select **OK** to perform the Gain Change or **Cancel** to cancel the operation.
7. Use the **Undo** function (Ctrl+Z) in the Edit menu if you are not happy with the results.

DC Filter

The DSP function removes the DC component from a sample, centering the waveform around the zero axis.

► To DC Filter a Sample

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. Usually, you will DC filter the entire sample. In this case you don't have to select a specific area.
 - Selecting a specific region of the sample to DC Filter is not recommended as this can create a sudden thump or pop at the selection point from the sudden change in the waveform.
3. Press the **DC Filter** button. The sample will be centered around the zero axis.
4. Use the **Undo** function in the Edit menu if you aren't happy with the results.



Reverse

Reverse flips the selected portion of the sample so it plays backwards.

► To Reverse a Sample

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. **Magnify** the waveform using the Zoom button below the display.
3. **Select the area** of the waveform you want to reverse.
4. If the sample is stereo you will be given the option to reverse the right side only, left side only or both.
5. Press the **Reverse** button. The selected section of the sample will be reversed.
6. Use the **Undo** function (Ctrl+Z) in the Edit menu if you are not happy with the results.

■ Press the Play button to hear the sample as you adjust the selection area.

Swap L-R Channels

Swaps sides of a stereo sample or moves a mono sample to the other side.

► To Swap the Left and Right Channels

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. Usually, you will select the entire sample. In this case you don't have to select a specific area.
Selecting a specific region of the sample to Swap can create a sudden thump or pop at the selection point from a sudden gain increase or decrease.
3. Press the **Swap L-R** button. The left and right channels will be swapped.
4. Use the **Undo** function (Ctrl+Z) in the Edit menu if you are not happy with the results.

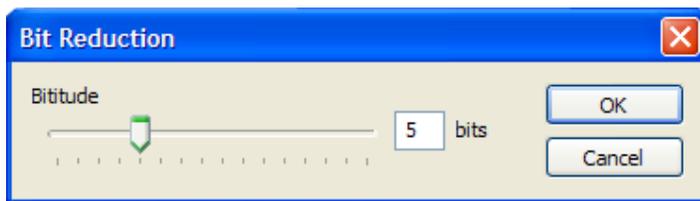
Bit Reduction

This DSP tool takes your pristine samples and transforms them into any bit resolution you select. Ever wonder what a 2-bit sample sounds like? Find out using the bit reducer. Bit values above eight will generally result in increased noise without much other audible difference.

After Bit Reduction, a sample will be stored as a 16-bit linear sample. This allows the tool to be used to convert 24-bit samples to 16-bits, with a resulting 1/2 reduction in disk storage (24-bit samples are stored on disk using 32-bits).

► To Reduce the Number of Bits in a Sample

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. **Magnify** the waveform using the Zoom button below the display if necessary.
3. **Select** the area of the waveform you want to Bit Reduce. Press the Play button to hear the sample as you adjust the selection area.
4. Press the **Bit Reduction** button. The following dialog box appears.



5. Select the **number of bits** you want.
6. Select **OK** to perform the Bit Reduction or **Cancel** to cancel the operation.
7. Use the **Undo** function (Ctrl+Z) in the Edit menu if you aren't happy with the results.

Time Compression

This DSP function allows you to change the length of a sample without changing the pitch. This can be extremely useful for fitting samples to the beat of a song or fitting dialog into a spot. Samples can be compressed or expanded in length from 50% to 200% in steps of 1/10th of one percent. Stereo samples maintain phase accuracy when time compressed or expanded.

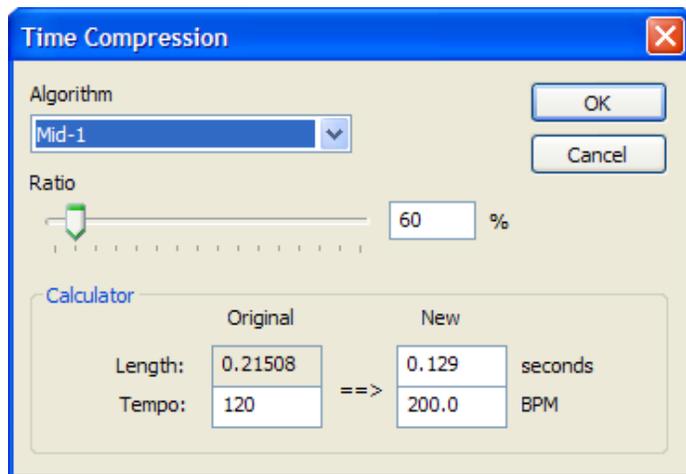
Large time compression ratios can produce undesirable artifacts in the sample. Stick with small ratios on critical audio parts. The artifacts produced by this processor can be used to create weird special effects. Try stretching-compressing-stretching-compressing, etc. if you want to really mess up a sample.

The tempo calculator makes it easy to change the tempo of sampled grooves. Simply set the original tempo to the tempo of your groove, then adjust the ratio until the new tempo matches the desired tempo.

- **Time Compression Ratios** - 200% = Double the Length
50% = Half the Length

► To Time Compress a Sample

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. **Magnify** the waveform using the Zoom button below the display if necessary.
3. Select the area of the waveform you want to Time Compress. Press the Play button to hear the sample as you adjust the selection area.
4. Press the **Time Compress** button. The following dialog box appears.



5. Select the **Time Compression/Expansion ratio**. The calculator shows the original length and tempo as well as the new length and tempo after processing.
6. Choose the **processing algorithm** using the table below as a guide.
7. Select **OK** to perform the Time Compression/Expansion or **Cancel** to cancel the operation.
8. Use the **Undo** function (Ctrl+Z) in the Edit menu if you aren't happy with the results. You may have to try a couple of algorithms before you get the best quality.

The following table lists the algorithm choices.

deep	Predominant bass (to 20 Hz)
bass	Predominant bass (to 20 Hz)
mid-1	Average source material
mid-2	Average source material (high-mids)
high	Source material with high frequencies
tight	Maintains time accuracy - Drum loops
broad	Low bass energy but critical highs
broad-smooth	Both high & low freq. - Smooth output
difficult	Inharmonic or broadband material
noisy	Non-pitched - Sound Effects, etc.
tight-smooth	Rhythmic accuracy - Use small ratios
x-smooth	Rhythmic accuracy - Use small ratios

Pitch Shift

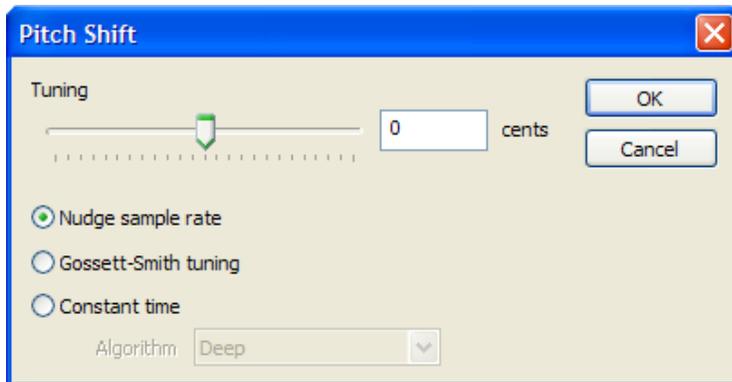
Pitch Shift offers three different ways to change the original pitch of a sample.

- **Nudge Sample Rate** - Changes the sample rate to vary the pitch. The higher the pitch, the lower the sample rate.
- **Gosset-Smith Tuning** - A high-quality, constant sample rate interpolation method.
- **Constant Time** - These algorithms change the pitch, without changing the length of the sample. Constant time pitch change may introduce unwanted artifacts into the sample, especially when large intervals are selected.

The maximum pitch shift amount is: ± 1200 cents (100 cents = 1 semitone).

► To Change the Pitch of a Sample

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. **Magnify** the waveform using the Zoom button below the display if necessary.
3. **Select the area** of the waveform you want to pitch shift. Press the Play button to hear the sample as you adjust the selection area.
4. Press the **Pitch Shift** button. The following dialog box appears.



5. Select the **Tuning** change in cents.
6. Select the **type of pitch shift** you want to perform: Nudge Sample Rate, Gosset-Smith, or Constant Time.
7. If you chose "Constant Time" pitch shift, select one of the processing algorithms using the table below as a guide.
8. Select **OK** to perform the pitch shift or **Cancel** to cancel the operation.
9. Use the **Undo** function (Ctrl+Z) in the Edit menu if you aren't happy with the results. You may have to try a couple of algorithms before you get the best quality.

The following table lists the algorithms for constant time pitch shift.

deep	Predominant bass (to 20 Hz)
bass	Predominant bass (to 20 Hz)
mid-1	Average source material
mid-2	Average source material (high-mids)
high	Source material with high frequencies
tight	Maintains time accuracy - Drum loops
broad	Low bass energy but critical highs
broad-smooth	Both high & low freq. - Smooth output
difficult	Inharmonic or broadband material
noisy	Non-pitched - Sound Effects, etc.
tight-smooth	Rhythmic accuracy - Use small ratios
x-smooth	Rhythmic accuracy - Use small ratios

Transform Multiply

This powerful synthesis tool multiplies two signals together in a way that the output signal contains the timbral and time elements of both inputs. This process, which is also known as *Convolution*, reinforces frequencies common to both sources and discards frequencies not present in both sources. Because of this characteristic, Transform Multiplication works best with sounds that are harmonically rich.

Convolution has recently become popular as a way to model natural sounding reverberation from an actual room. You can experiment with this yourself by recording an impulse response in a reverberant space and then Transform Multiplying it with a sound of your choice. To create a simple impulse response, record a starter pistol or sharp percussive noise burst in your favorite room. If you're interested in this subject, you'll find information about convolution and free impulse responses on the Internet.

To use Transform Multiply, just copy a sample region to the clipboard, then choose another sample and Multiply!

◆ The output of any filter is the convolution of the input signal with the filter's impulse response.

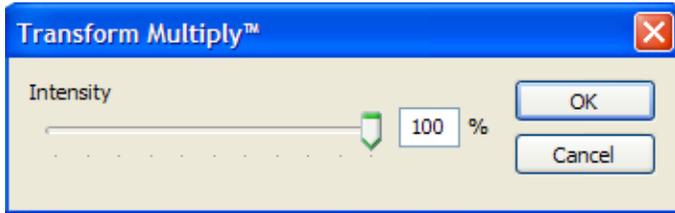
Ideas for Using Transform Multiplication

- Reverb impulse responses will add their reverb signature to any sample.
- Percussion transients make good source material.
- A stereo impulse response can be T.M.'ed with a mono source to create a stereo output.
- Apply pitch shift or time compression to room impulse responses to change the resonance and decay time of the result.
- Reversing the impulse (clipboard sample) will reverse the attacks of transients in the main sample.
- Make your own impulses. Select a sample region, fade it, then copy to the clipboard and Transform Multiply
- Using speech as one of the sources, it is possible to "speak from within" another instrument (violins, bassoons, cymbals, etc.).
- Splicing silence to the beginning or ending of sounds will change the spectral characteristics of the result.

■ The E-MU Transform Tool Kit sound bank contains samples designed for use with Transform Multiplication.

► **To Transform Multiply**

1. Select a **harmonically rich sample**, or short region of a sample and **copy it to the clipboard** (Edit Menu, Copy or Ctrl+C).
2. Select **another sample** and highlight the region you wish to process.
3. Select **Transform Multiply** from the Sample Edit menu or by clicking the tool button.  The following pop up dialog box appears.



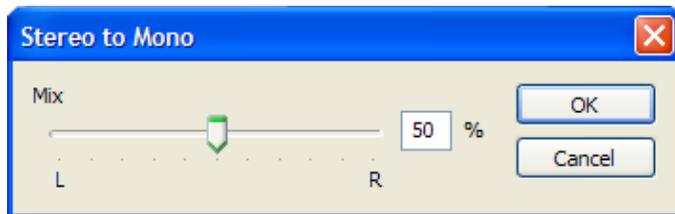
4. Intensity sets the ratio of effect to original sound. (In this case, the original sound is the second sample you chose, not the one on the clipboard.) For radical sound transformation set this value to 100%. To perform reverb convolution, choose a somewhat lower value such as 20-30%.
5. Press **OK** to Transform the two sounds.

Convert to Mono or Stereo

Converts a stereo sample to mono or a mono sample to stereo. Converting to stereo duplicates the sample on both sides and slightly detunes them.

► **To Convert between Mono and Stereo**

1. **Click once on a sample icon** in the Tree to select it for editing. The waveform display appears.
2. This function only operates on the entire sample so there is no need to select a region.
3. Press the **Convert Mono-Stereo** button. When converting mono to stereo, the sample is simply duplicated. When converting from stereo to mono, the following dialog box appears.



4. **Adjust the mix of left and right signal** in the new mono sample.
5. Select **OK** to perform the conversion or **Cancel** to cancel the operation.
6. Use the **Undo function (Ctrl+Z)** in the Edit menu if you aren't happy with the results.

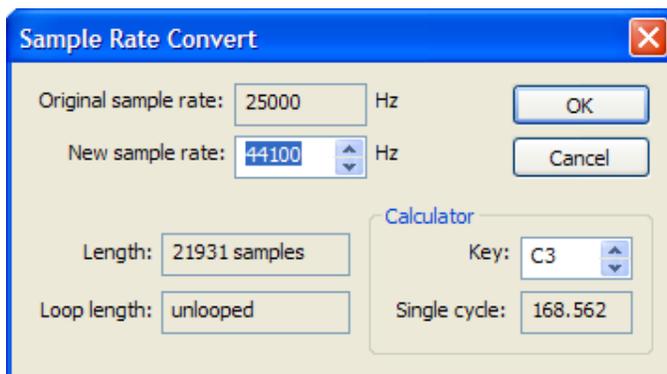
Sample Rate Convert

Sample Rate Conversion digitally changes the sample rate from 8000 Hz to 100 kHz. Sample Rate Conversion can be used to match the sample rate for use in other applications or to decrease the memory size of a sample.

Sample Rate Conversion can also be used for exactly matching the sample rate to a multiple of the sound's frequency for perfect single cycle loops. Set the built-in sample calculator to the note you want to loop and it automatically calculates the number of samples in a single cycle loop. Next, adjust the sample rate until you get close to an integer value for the single cycle loop. Set the loop length to this value when you go to loop it.

► To Change the Sample Rate

1. Click once on a **sample icon** in the Tree to select it for editing. The waveform display appears.
2. You can only change the sample rate of the entire sample so there is no need to select a region.
3. Press the **Sample Rate Convert** button. The following dialog box appears.



4. Select the **new sample rate**. The sample's new length is displayed.
 - If you want to create a single cycle loop, set the calculator's Key field to match the pitch of the sample and adjust the new sample rate until you get an integer number. Use this value as the loop length when you loop the sample.
5. Select **OK** to perform Sample Rate Conversion or **Cancel** to cancel the operation.
6. Use the **Undo** function (Ctrl+Z) in the Edit menu if you aren't happy with the results.

Xtractor

The Xtractor is a new audio manipulation tool that can cut or boost audio levels based on stereo position and frequency. If a song was mixed using stereo panning, this tool allows specific instruments or vocals to be removed, isolated, or pitch shifted. The Xtractor needs a stereo sample to work correctly since it uses the stereo position to isolate audio parts.

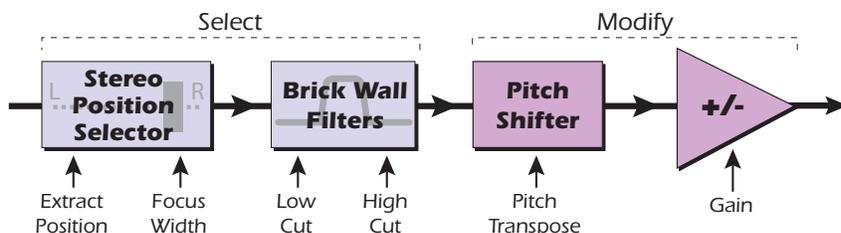
Applications

- Vocal elimination/vocal isolation
- Change the gain of specific instruments within a complex mix (*remix a song*)
- Correct out-of-tune vocal parts in a completed song
- Isolate a specific instrument or vocal for processing, then return it to the mix
- Mix analysis - find out how a song is mixed
- Unique special effects creation

► Using the Xtractor typically involves these easy steps:

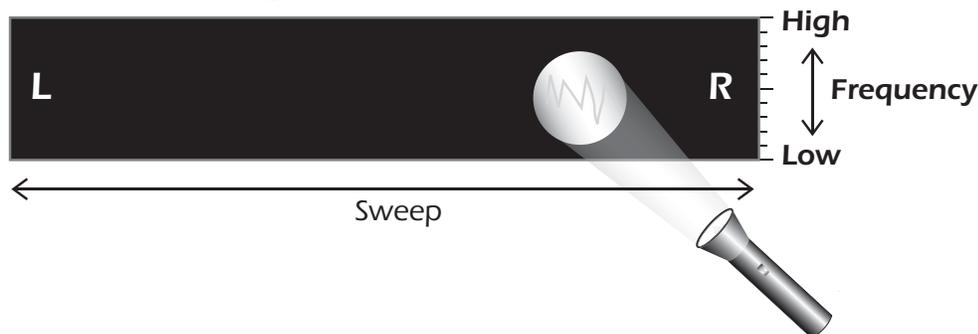
1. Select an appropriate stereo sample (typically, complete songs with several instruments or vocal parts panned across the stereo field.)
2. Audition using the Position and Frequency controls to determine the stereo position and frequency range of the element you wish to extract.
3. Use the Gain control to isolate, eliminate or change the gain, or use the Pitch control to shift the pitch of the selected element.

Xtractor Block Diagram

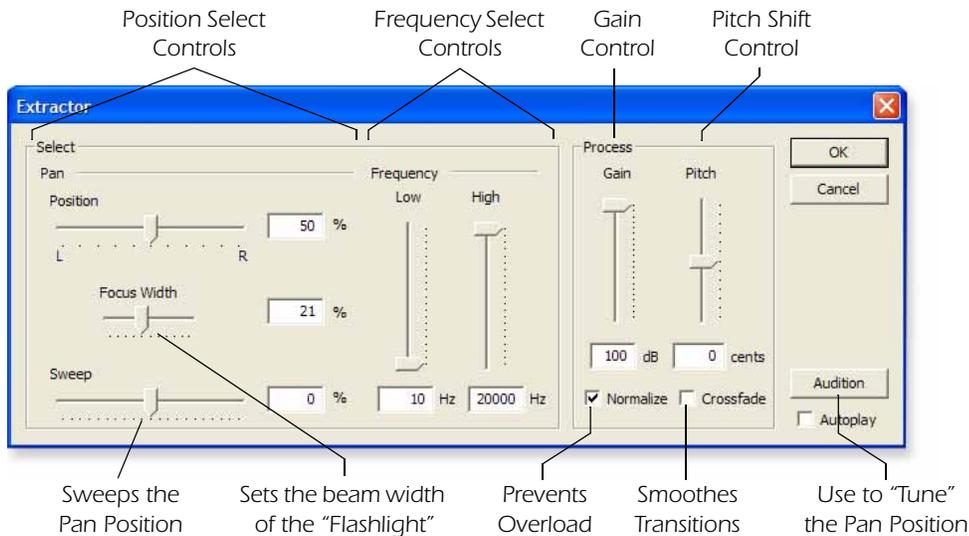


The Xtractor allows you to select a stereo position and frequency range and then change the pitch or gain of this selected range.

Complete Audio Mix

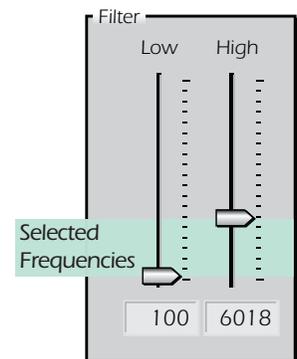


A flashlight sweeping across the stereo mix is a good analogy for what the Xtractor does. It highlights audio information based on stereo position and frequency.



Xtractor Controls

Control	Function
Position to Extract	This control selects the center point of the stereo position you wish to gain change. 0% = full Left; 50% = Center; 100% = full Right
Focus Width	This control sets the width of the Position to Extract region. The focus width is adjustable from 1% to 32%. With the Position to Extract set to 50% and the width of 20%, the Extract Region would include instruments panned from 40% to 60% around center.
Audition	Turn Audition on, set the Gain to 100%, then experiment with the Selection controls to find the audio you wish to extract.
Autoplay	This feature automatically turns Audition on when Extract is first selected.
Filters	After the desired stereo position has been selected, the filters are used to zero in on the desired material. Frequencies BETWEEN the slider settings will be selected for gain adjustment.
Filter Low	This is an extremely steep highpass filter adjustable from 0 to 9999 Hz. Only the frequencies ABOVE this setting will be selected.
Filter High	This is an extremely steep lowpass filter adjustable from 0 to 20 kHz. Only the frequencies BELOW this setting will be selected.
Gain	This important control allows you to apply subtle or extreme gain changes in the selected stereo position and frequency range. A setting of +100dB extracts the selection; -100dB eliminates the selection.
Pitch Change	This control allows you to change the pitch of the selected stereo position and frequency range. The range of the pitch shifter is ± 1200 cents (± 1 octave) in 1 cent steps.
Normalize	This feature prevents the sample from being clipped when extreme amounts of Gain are applied. When using small Gain amounts, you might want to turn this function off.



Frequencies between the slider settings will be selected.

Xtractor Controls

Control	Function
Crossfade	A slight crossfade is normally applied at the edges of a processed section of audio to prevent abrupt tonal shifts. This box should normally be checked (on). When working with audio loops, turn crossfade off.
Sweep	This function is intended primarily as an analysis tool, but can also create unusual special effects. Sweep automatically adjusts the Position to Extract control over the course of the selected sample region. For example, with the Position to Extract control set to 0% and with Sweep set to +100%, the Position to Extract will sweep from full left to full right as the sample plays.

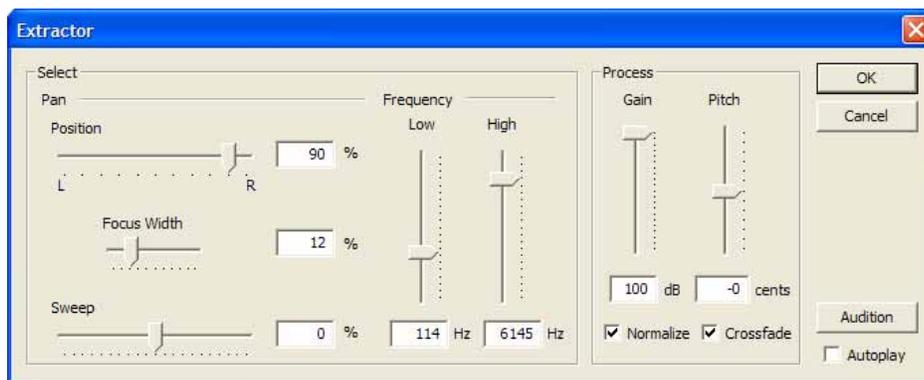
Vocal or Instrument Isolation/Elimination

The Xtractor is the perfect tool if you want to isolate a particular instrument or vocal from the mix.

First, you need to find out where the audio is located in stereo and frequency space. With **Audition** on and the **Gain** set to 100% adjust the **Position**, **Width** and **Frequency** controls until you hear the audio you want being isolated. (Make sure that the Sweep control is set to 0 - no sweep.) Because the Gain is set to +100 dB, the audio that falls within the parameters will be boosted higher than everything else.

- The *Normalize On* checkbox prevents the signal from clipping because of the extreme gain change.
- The *Crossfade On* checkbox smooths the transition between processed and unprocessed audio sections.

■ You can also try to locate the audio you want to isolate by using attenuation (a negative gain value). Instead of listening for a boost, wait for the audio to disappear as you adjust the Width and Frequency controls.



► To Isolate selected Audio from a Mix

1. If Autoplay is on you'll hear audio as soon as the Xtractor window appears. If Autoplay is off, press the **Audition** button.
2. Set the **Gain** to 100dB and make sure **Sweep** is set to 0%.
3. Adjust the **Position**, **Frequency** and **Focus Width** controls for the best possible isolation of the instrument you want.
4. To bring up the instrument in the mix, use less extreme Gain change settings (+6 dB to +12 dB). Turn off **Normalize After** for small gain changes.
5. Press **OK** and listen to the file.
6. **Undo** and try again if you're not satisfied with the result.

► **To Eliminate selected Audio from a Mix**

1. Follow the instructions above, except set the **Gain to -100 dB**.
2. To bring down the instrument in the mix, use less extreme Gain change (-6 dB to -12 dB). Turn off **Normalize After** for small gain changes.

► **To Pitch Shift selected Audio in a Mix**

1. Follow the instructions for isolating audio, except change the pitch instead of changing the gain after you have isolated the desired audio part.
2. Turn **Normalize Off**. If the pitch shifted audio is lower in volume, you may want to boost the Gain (using the Gain control) by a few dB.

■ **Tip:** Changing the pitch by a few cents can create add chorus or flange effects to the selected instrument.

► **To Process a Vocal and then Return it to the Mix**

This trick allows you to completely remove a vocal (*or other element of a mix*), completely reprocess it, and then return it to the mix. This technique is perfect for Mash-ups and is loads of fun!

1. **Make TWO copies** (Duplicate) of your sample.

Start work on Duplicate #1 - Isolate the Vocal

2. Turn **Normalize on**.
3. Crank up the **Gain to +100 dB**.
4. **Find the vocal** using the Pan control (narrow the focus using Low and High Filter e.g. to avoid bass guitar / kick drum / cymbals).
5. **Press OK** to process, leaving only the vocal in #1.

Start work on Duplicate #2 - Remove the Vocal

6. Leave all the controls the same, except...
7. Turn down the **Gain to -100 dB**.
8. **Press OK** to process it, removing the vocal, leaving everything else in #2.
9. Modify the vocal in sample #1 to meet your liking.

Put it back together

10. Select All (of #1), and **Copy** to clipboard.
11. Go to sample #2.
12. Move cursor to start of sample.
13. **Paste**, using Mix Mode / Equal Power.

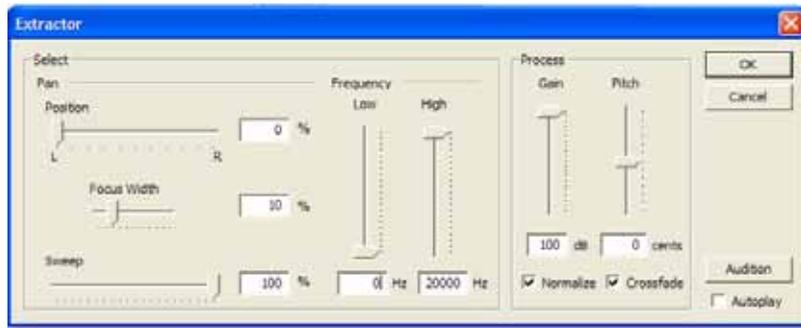
A Visual Mix Analysis

A visual mix analysis allows you to visually plot the position of each instrument as the cursor sweeps across the sample section.

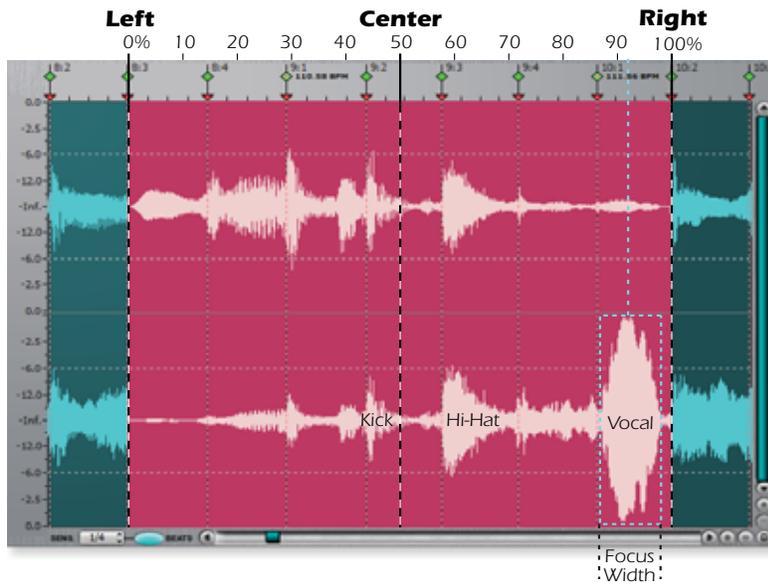
► To Analyze a Sample

1. Select a stereo sample in the Sample Edit window.
2. Magnify the waveform using the Zoom button below the display if necessary.
3. Select the region of the sample you wish to analyze. (Choose a short section for your initial experiments to shorten processing time.)
4. Press the **Extract** button. The Extract dialog box appears.
5. Set the controls as shown below and click **OK** to analyze the selected section.

**Visual
Analyze
Mode
Settings**



6. Press **Play** (spacebar) and listen carefully as each instrument jumps out as the cursor sweeps across its position in the mix. Visualize the sample section as a diagram showing the stereo positions of each instrument. In the example below, the Vocal is positioned at about 92% to the right and its Focus Width is about 12%.



After analysis, playing the sample section allows you to see and hear the exact stereo position of each element of the mix.

7. After you have identified the position of the instrument or vocal you want to process, press **Undo** (Ctrl+Z) to return the section back to normal.

IMPORTANT NOTE about the Region Page and Editing Samples

If you're using the Regions feature in the Voices and Zones window, you should keep the following in mind:

If a region contains only a part of the sample, DO NOT edit the sample in the Sample Editor (e.g. Time Stretch, Sample Rate Convert, etc.). Doing so will cause region information to become incorrect. This does not apply if a region contains the entire sample.

Since regions contain the entire sample by default, this is only a potential issue if you have created a region. To ensure a region contains an entire sample, you can always Alt-click the sample region bar to restore the region to its default setting.

For more information on regions, [see "Region Page" on page 115.](#)

Batch Processing

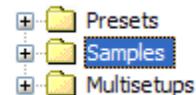
You can run most of the sample edit tools on multiple samples at once. This includes Export and External Edit.

You can perform the following operations on multiple samples at once:

- Reverse
- Adjust Gain
- Convert to Mono/Stereo
- Normalize
- Swap Channels
- DC Filter
- Convert Sample Rate
- Time Compress/Expand
- Pitch Shift
- Bit Reduction
- Transform Multiply
- Beat Analysis
- Export
- External Edit

► To Batch Edit a Group of Samples

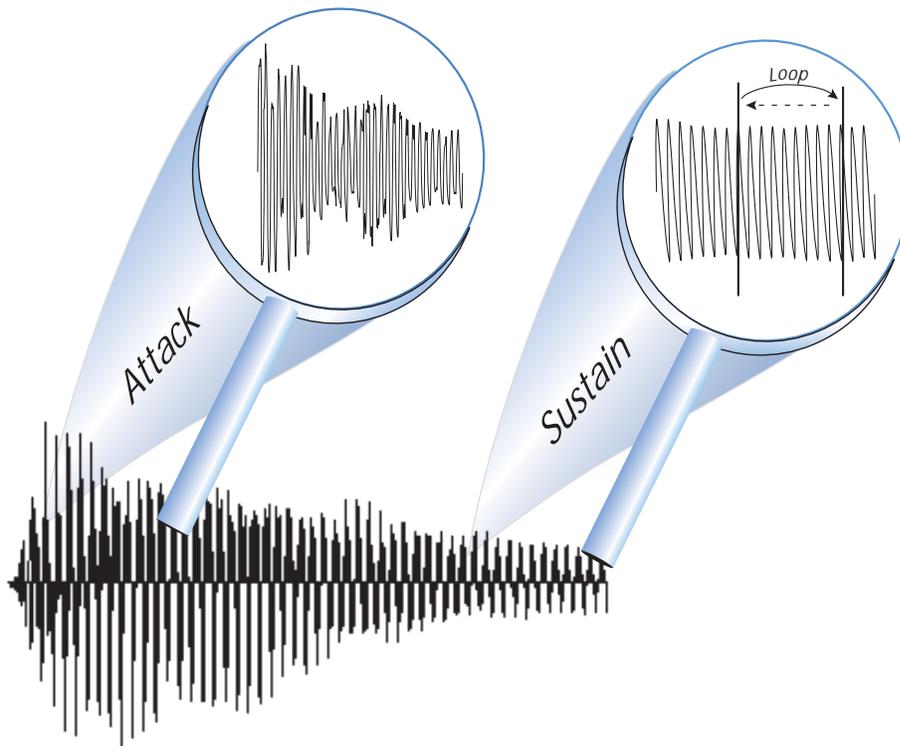
1. From the Sampler Tree on the left, click the **Samples** folder.
2. In the pane to the right, select the samples you want to process. Hold the **Shift** key while clicking to select contiguous samples, or hold the **Ctrl** key while clicking to select multiple non-contiguous samples.
3. Either:
 - Click the operation in the toolbar that you want to perform on the selected samples.
 - From the **Samples** menu, select the operation you want to perform on the selected samples.
 - Right-click the selected samples, and select the operation you want to perform on the selected samples.
4. If the operation requires more information, the appropriate dialog appears. Whatever settings you choose affect all the selected samples. For more information on each operation, see [“Sample Edit Tools” on page 185](#).



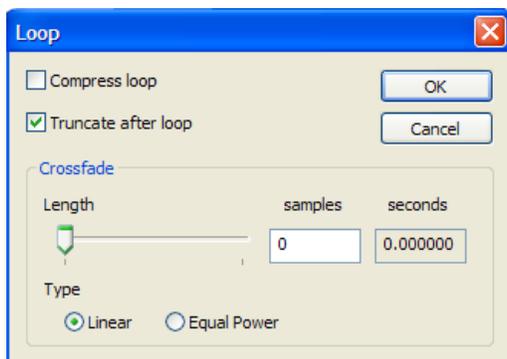
Looping and Loop Processing

Looping is a technique that can extend a sound's length indefinitely. It is based on the fact that many sounds start off with a complex attack transient, then settle down to a comparatively steady sound.

Although an Emulator X sample can have multiple loops, most sampled instrument presets only use a single loop. Multiple loops are generally used for special effects or for the TwistaLoop function as described on [page 209](#).



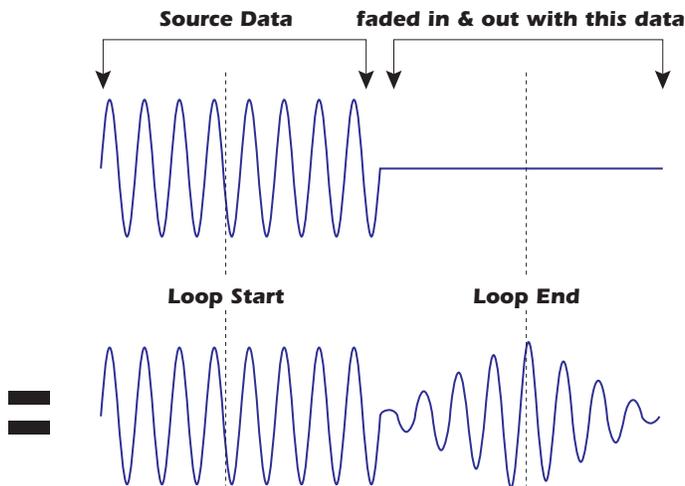
For an aural example of how looping works, listen carefully to a plucked guitar string. The first part of the sound consists of a complex mixture of pick noise and several harmonics; then the string quickly decays down to a steady repetitive tone. Since the latter part of the waveform is repetitive, you can mark off a loop of the repetitive section, and play that looped section for as long as the key is held down. After playing to the end of the loop, the sound jumps back to the beginning of the loop and plays through the loop again. This process repeats until you release a key. If the "Loop-in-Release" function is On, the sound continues to loop until the volume dies away.



Crossfade Looping

Crossfade Looping fades between the beginning and end of the loop so that as the end fades out, the beginning fades in. This virtually eliminates the clicks and pops that can occur with other types of looping. Sound data from around the loop start point is faded into the sound data around the loop end point so that the data at those loop points is identical.

The maximum allowable crossfade length is based on the distance from the start point of the sample. (Move the start point toward the end of the sample to allow a longer crossfade.) You can select linear crossfade or an equal power curve.



In the diagram above, a sine wave is crossfaded with silence to show the fade-in and fade-out. In an actual crossfade loop, both signals would be crossfaded together.

Truncate after Loop

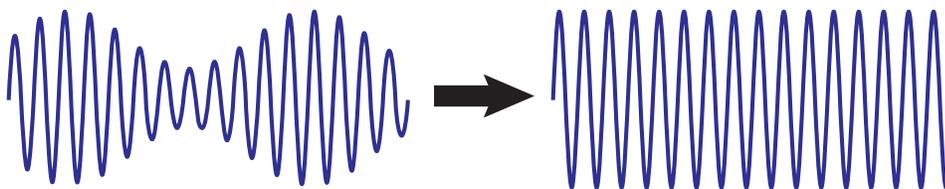
Discards any information after the loop end point in order to conserve memory.

Loop Compression

The looped section of the sample can be compressed, which “evens out” any changes in amplitude. Changes in amplitude can cause “breathing” effects, thumping or clicks.

Amplitude Variations in Loop

After Loop Compression



- **Crossfade Size:** (Crossfade Loop) This function smoothly crossfades between the waves on either side of the loop so that the waveform at the loop point is identical. Since Crossfade Looping alters the sound data, you may notice timbral or amplitude variations. A size of 0.00 disables crossfade looping.
 - **Crossfade Type - Equal Power:** This is a weighted crossfade that produces no apparent level shift, and is the commonly used mode.
 - **Crossfade Type - Linear:** This provides a straight mathematical crossfade. Use linear mode with samples whose splice points are already close to optimum, such as a sample that has already been auto correlated.

■ Use larger crossfade loop sizes with complex sounds; shorter sizes when you simply want to mask a slight tick in the loop.

Loop Settings

These controls allow you to turn looping On or Off, select whether the Loop in Release function is On or Off, select the current loop, lock the loop length, turn triplet playback On or Off and dial in the selection start and end points with sample accuracy. There is also an overall tempo display.



Loop Button

This button enables or disables looping. You'll be happy to know that if you loop a sample and turn the loop off, the loop pointers remain exactly where you set them when the loop is turned back on (even after saving).

Loop in Release Button

With Loop in Release off, the sample stays in the loop as long as the key is held, and plays the sound data after the loop when the key is released. With Loop in Release on, the sample remains looping when the key is released.

Loop Select

This field selects the current loop when multiple loops are defined. For more information about multiple loops see ["Using Loops" on page 217](#).

Lock Button

The Lock button fixes the loop length at its current value. This is a handy feature when you have found the correct loop length and are looking for a better start point. The lock button is also very useful for creating groove loops and single cycle loops.

Select Loop

This tool automatically selects the region between the loop markers for the currently selected loop. Select Loop is available from the toolbar button  or by selecting it from the Sample Edit pull-down menu.

Select All

This tool automatically selects the entire sample. Select All is available from the toolbar button  or by selecting it from the Sample Edit pull-down menu.

No Snap

Selections made in the Sample Edit window will not snap to any boundaries when this option is active. No Snap can be enabled from the toolbar button  or by selecting it from the Sample Edit pull-down menu.

Snap To Marker

Selections made in the Sample Edit window will snap to Loop Markers, Note-Onsets, or Beat Markers when this option is active. Snap To Marker can be enabled from the toolbar button  or by selecting it from the Sample Edit pull-down menu.

Snap To Zero-Crossings

When this function is on, sample selections made by dragging or clicking in the waveform display will automatically snap to the nearest zero-crossing points when the mouse button is released. For stereo samples, the zero-crossing is determined by the nearest zero-crossing on either channel. See also [“Zero Crossing” on page 208](#).

Turn this function on, either by pressing the Toolbar button  or by selecting it from the Sample Edit pull-down menu.

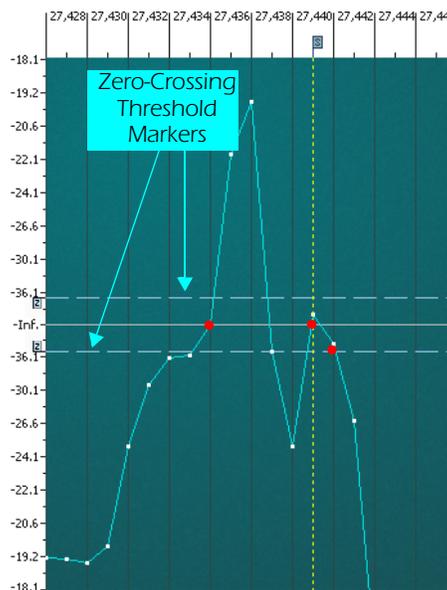
Adjustments to the selection using the keyboard, and menu command selections are not affected by this switch. The switch holds its setting for each sample.

Background: Zero Crossing Threshold

The Zero Crossing Threshold determines how close to zero a sample has to be to qualify as a zero crossing. This allows more points to be selected as zero crossings.

To qualify as a zero crossing, the wave must have passed through the zero line since the last selected zero crossing and the sample must lie inside the zero crossing threshold.

In the example on the right, the samples marked in red will be selected as zero crossings. See also [“Zero Crossing” on page 208](#).



► To Change the Zero Crossing Threshold

1. Enable “Show Zero Crossing Threshold” from the Display menu, or by pressing the toolbar button. 
2. Press the Vertical Zoom button until you can see the Zero Crossing Threshold Markers on the left side of the waveform display.
3. Drag the markers apart to select more zero crossings. Place them closer together to exclude more zero crossings.

Delete Unused Samples

When creating a custom bank, you may end up with a bunch of unused samples which are not used in any of the presets you created. This utility allows you to discard the unused samples from the bank.

► To Delete Unused Samples from the Bank

1. Right-click on the sample folder in the tree.
2. Select **Delete Unused Samples**.
3. Any samples not used in any preset will be erased from the bank.

▼ **Caution:** Delete Unused Samples erases the actual sample data. Make sure your original samples are backed-up before using this function.

How to Loop a Sample

Looping has gained a reputation for being something of a black art. In actuality, it's really not that difficult to loop most sounds, especially with all the looping tools provided by the Emulator X.

The Attack portion of the sound is the beginning portion of the waveform. The attack is usually composed of changing harmonics and is difficult to loop. To get a smooth sounding loop, move the loop start and end points into a portion of the wave where the sample looks and sounds repetitive and is steady in amplitude. Changes in amplitude between the loop start and end points will be heard as sharp clicks or pops. You don't want this.

1. Select a **sample** from the Tree by double-clicking on the sample name or icon.  The Sample Editor window appears. You can also select samples by double-clicking on the sample name in the Voice Processing window, or by selecting a sample and choosing "Open Sample" from the Preset menu.
2. Press the **Select Loop** button to jump to any pre-existing loop.
3. **Magnify** the waveform using the Zoom button if necessary. Look for a region of the sample near the end where the wave looks repetitive.

Initial Settings

4. Place the **Loop End** marker near or at the end of the sample by dragging it into position. Position it at a zero crossing by holding the Right Control key and pressing the left or right arrow keys. (See [page 206](#) for more information.)
5. Use the **Autocorrelate Loop** (Ctrl + K) feature in the Sample Loop menu to get you into the ballpark.

Listen & Adjust the Start Point

6. Make sure **Loop Play** in the transport toolbar is turned off and select **Play**. The sample will play and continue playing within the loop boundaries.
7. While the sample is playing, slowly **drag the Loop Start** marker forward and back through the sample. Notice how the character and pitch of the sound changes. Try to match the pitch of the loop with the pitch of the sound. This will give you an idea of the proper loop length. Try to find a point where the loop sounds "smoother" with less "ticking".
8. When you get just a soft "tick", try holding the mouse over the increment and decrement buttons on the loop controls above the waveform display. This will move the loop one sample at a time.



9. Try **Autocorrelation** again. Autocorrelation only moves the Start point, so if it doesn't find a good loop, move the Loop End point slightly and try it again.

Can't find it?

10. Repeat step 7, moving the Loop End marker this time. Visually examine the end of the waveform and try to find an area that looks similar, then move the markers around in that area.

▼ **Note:** Some sounds truly cannot be seamlessly looped. Percussion sounds and ensembles are notoriously hard to loop.

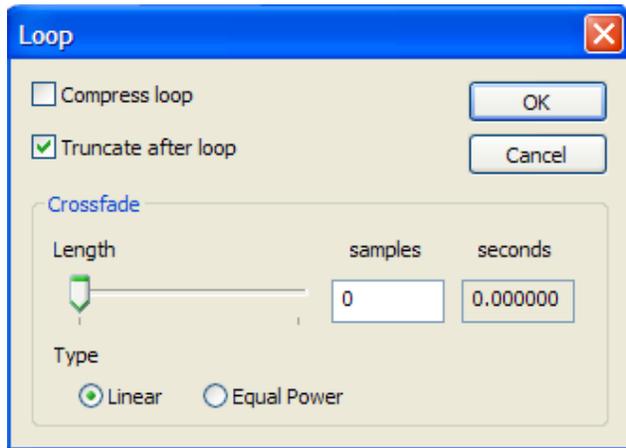
If you're just learning to loop, practice with an easy sound such as your voice saying "Ahh".

■ Looping is an art which takes a bit of practice to learn. Auto Correlate works best when you have already found a fairly good loop point by ear.

► **Crossfade Looping & Loop Compression**

Use these tools when you have a loop that's good, but not great. Since these tools work by changing the sample data, it's best to use them only when necessary. Crossfade looping is a powerful tool in your arsenal, but like most things, it takes a bit of practice to master.

1. With your loop points set as good as you can get them, select **Loop Processing** from the Sample Edit menu or from the toolbar. The following dialog box appears.



- Use a larger crossfade loop length with complex sounds and shorter lengths with when you just want to mask a slight tick in the loop.

2. The maximum allowable crossfade length is based on the distance from the start point of the sample. What length should you use? It all depends on the sound. Begin with a long crossfade and use the Undo function (Ctrl + Z) if you don't like what you're hearing. Choose **Equal Power**. **Truncate after Loop — On**.
3. If you get a nice timbral crossfade but still have amplitude variations in the loop, turn **Compress Loop On**.

- Move the start point toward the end of the sample to allow a longer crossfade.

Zero Crossing

The term *zero crossing* refers to the point at which the positive slope of a waveform passes through zero. When looping it's sometimes useful to locate zero crossings in order to make glitch-free loops. Both channels are used for zero-crossing selection when working with stereo samples.

► **To Move the Loop Markers to Zero Crossings**

1. Hold the **Left Ctrl** key and press either the left or right **Arrow** keys to move the **Loop Start Point**.
2. Hold the **Right Ctrl** key and press either the left or right **Arrow** keys to move the **Loop End Point**.

- The left arrow moves the loop marker toward the sample start.

The right arrow moves the marker toward the sample end.

8 - TwistaLoop

Introduction

TwistaLoop™ is a tool that automatically analyzes any audio by dissecting it into its rhythmic components. This allows you create and edit multiple loop points and regions, and have total control over tempo using real-time audio time expansion/compression.

You can playback analyzed samples at a custom speed, or in time with Emulator X's clock (or an external clock source). Analyzing a sample with TwistaLoop™ consists of three easy steps:

1. Run Beat Analysis on the sample.
2. Verify that you achieved the desired results, and make any fine adjustments to the analysis.
3. Create as many loops and regions from the analyzed sample as you want.

Once you create loops or regions from an analyzed sample, you can do the following:

- Change the playback speed of any loop.
- Choose which loops are repeated when played back.
- Set up a continuous controller to change in real time which loop plays.
- Sync your loops to Emulator X's master tempo, or to an external tempo sources, such as a VST host.
- Play different beats or bars from the same sample.

Analyzing Beats In the Sample Edit Window

To run Beat Analysis on a sample, you'll need to start in the Sample Edit Window.

► To Get to the Sample Editor Window

- Click on any sample in the tree to view that sample in the Sample Editor window.
- Double-click on any sample in the "Voices and Sample Zones" window.

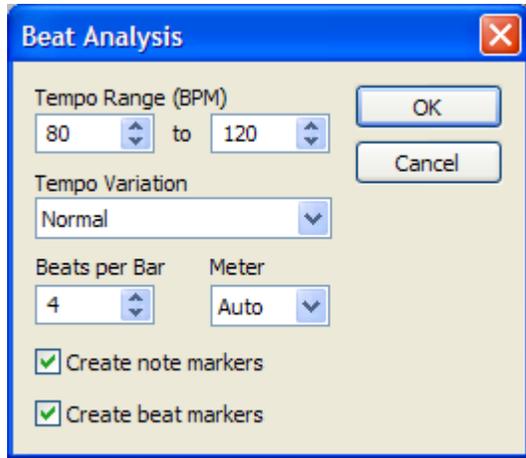
◆ For more information on using the Sample Edit window, see [Editing a Sample on page 169](#).

Running Beat Analysis

Beat Analysis looks at a sample and determines its rhythmic aspects. Once you analyze the sample, it's easy to create loops based on the Beat and Note Markers that Beat Analysis produces. This process is easy, and produces great accuracy.

► To Run Beat Analysis:

1. Select the sample you want to analyze and click anywhere in the sample display.
2. From the **Sample Edit** menu, select **Beats**, **Beat Analysis**, or press **Ctrl-b** on your keyboard. The following dialog appears:



3. Enter the Tempo Range, Tempo Variation, Beats per Bar, and Meter.
4. Click Create note markers and Create beat markers.
5. Click OK.

When analyzing, it helps to have some idea of the sample's original tempo and time signature. See below for a description of what these values mean.

Tempo Range (BPM)

This is the estimated tempo of the sample. You can enter values between 40 and 240 BPM (Beats per Minute).

Normally, beat Analysis correctly identifies a tempo. Beat Analysis is even more accurate if you have an idea of the sample's original tempo. In most cases, a range of 60 to 180 BPM is fine, but you can always narrow this value and re-analyze the sample.

When narrowing the value, try to choose a tempo range that encompasses your estimated tempo, but leaves room on either side. For example, if you think the tempo of your sample is around 100 BPM, you might select a range of 90 to 110.

■ **Tip:** 60 BPM is one beat per second, so if you have no idea what the original tempo is, you can at least use that as a guide.

Tempo Variation

Some samples are more rigid in tempo than others. Set the Tempo Variation depending on how well-timed your sample sounds.

Normal	For samples with a small to moderate variation in tempo.
Fast	Use for a sample that contains sudden or drastic changes in tempo.
Constant	Use for short samples with rigid tempos.

Beats per Bar

Sets the time number of beats per measure. You will probably want to use the number of beats per bar that the original sample used. However, you can achieve some interesting loops if you change this.

Meter

When analyzing a sample, shorter notes (or “pulses”) are recognized as either sixteenth note or triplet rhythms. The Meter setting determines how the analysis will treat these notes.

Auto	Automatically determines what type of rhythm the sample uses. If you analyze your sample and are dissatisfied with the results, try selecting Duple or Triple.
Duple	Treats shorter notes as straight 16th notes. Use duple to guarantee that all short notes will be recognized as 16th notes.
Triple	Treats shorter notes as part of a triplet. Use Triple for samples that have a shuffling rhythm.

Create note markers

Creates Note Markers when Beat Analysis is performed.

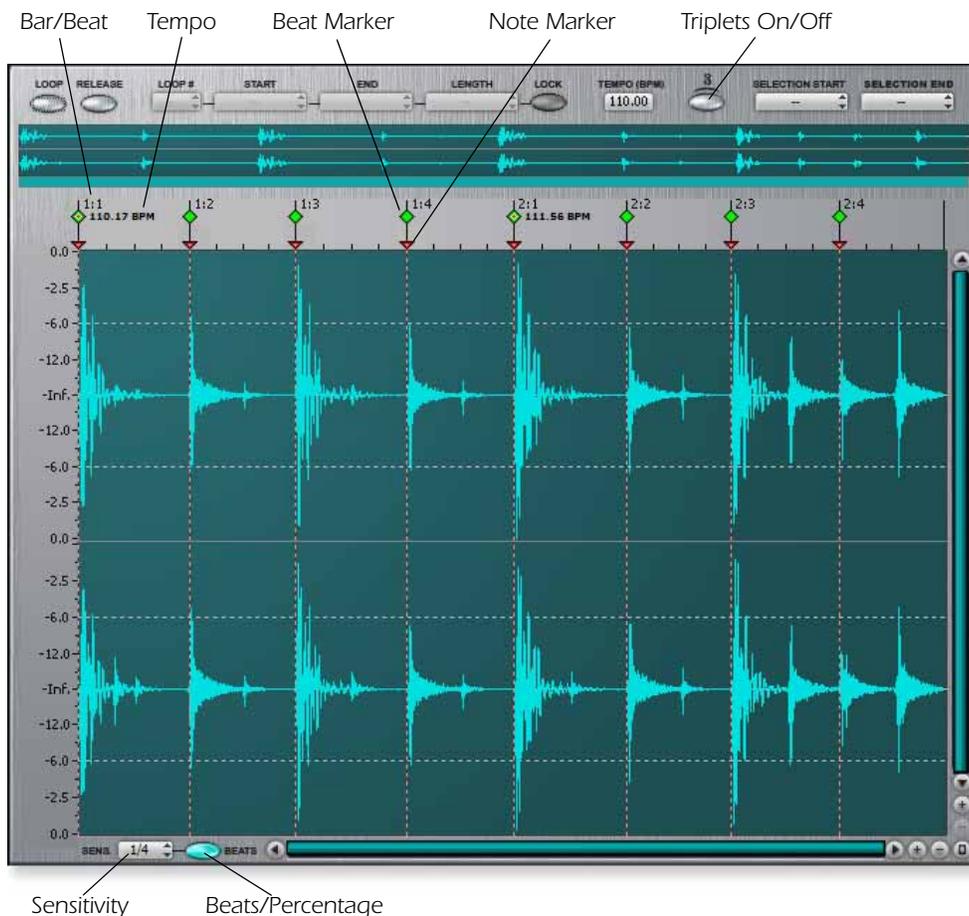
Create beat markers

Creates Beat Markers when Beat Analysis is performed.

Viewing Your Analyzed Sample

After you run Beat Analysis on your sample, Beat Markers  and Note Markers  appear on the ruler above the sample. The idea here is that Beat Analysis looks at your sample, and intelligently divides it into beats and notes. Each beat marker is assigned a bar number and beat number and depending on your tempo granularity settings, a tempo. The following diagram illustrates how this works:

■ Unless you are analyzing imported files, such as REX file, that already contain note onset markers, you should probably leave this box checked.



Try selecting an entire bar and playing it back. Make sure **Snap to Marker**  is selected. If you want to hear the bar loop, select **Loop Play**  from the toolbar or Transport menu.

If Beat Analysis was able to analyze your sample, you should hear a perfect loop of your sample. You can then create a loop from this bar to use as a voice. If you're unhappy with the way your sample was analyzed, you can run Beat Analyzer again, or you can fine tune the analysis manually.

Markers

After you analyze your sample, the horizontal ruler above the sample changes to the Bar:Beats scale. Beat Markers  and Note Markers  appear in the ruler, with the initial beat of each bar containing an orange center . You can turn the markers on or off by clicking the  buttons in the toolbar, or by right-clicking the ruler, and selecting **Show Note Markers** or **Show Beat Markers**.

Beat Marker Bar:  Beat: 	Beat Markers determine the location of each beat. When you create loops, make sure your loops begin and end on Beat Markers. This way, your loops properly sync to your tempo source.
Note Marker 	Note markers appear on note onsets determined by Beat Analysis. They allow Emulator X to determine where Beat Markers should go, and they work in the background to adjust your loops to the tempo you are using.
Bar/Beat 1:1	The Bar and Beat is shown next to each Beat Marker.
Tempo 105.56 BPM	Beat Analysis calculates the original tempo of a group of beats (by default one bar). When you change the tempo of a loop, Emulator X uses this tempo to correctly synchronize the loop to your tempo.

The Triplet Button

If your sample contains triplets and you want Emulator X to play them back as triplets when you change the tempo, click the Triplet button.



Sensitivity

The sensitivity setting determines the resolution of how Note Markers will be displayed. You can set the sensitivity in percentage from 0 to 100:



Or, if you click the **Beats** button, you can set the sensitivity ranging from eight bars intervals (8/1) to sixteenth note intervals (1/16):

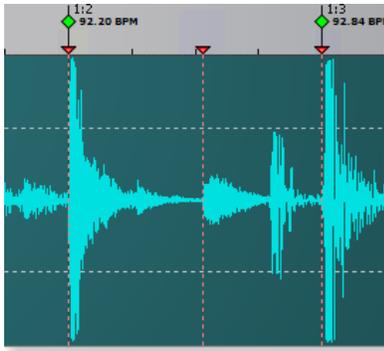


For example, if you set the sensitivity to 1/8, Note Markers fall on note onsets that appear near eighth notes. Similarly, setting the sensitivity to 1/16 causes Note Markers to fall on note onsets near sixteenth notes.

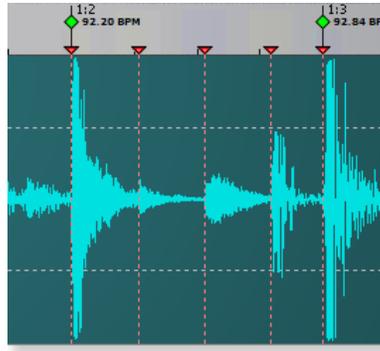
- A letter "t" next to a sensitivity setting indicates triplets.
- A letter "d" next to a sensitivity setting indicates duplets.

■ **Tip:** You can make specific Note Markers always appear, regardless of the sensitivity. Right-click the Note Marker, and select **Show Always**.

To reverse this setting, right-click the marker and uncheck **Show Always**.



1/8 Sensitivity



1/16 Sensitivity

Fine-Tuning Your Analyzed Sample

You can make manual adjustments once you've run Beat Analysis:

- Move the beat and note markers by clicking and dragging them.
- Add new beat or note markers by right-clicking the horizontal ruler, and selecting **Add Beat Marker** or **Add Note Marker**.
- To delete a beat or note marker, right-click the marker you want to delete, and select **Delete Beat Marker** or **Delete Note Marker**.

Shifting the Beats

Sometimes when you analyze a sample, Beat Analysis might confuse the first beat of a bar, resulting in bar numbers that are a fraction of beat off to the right or left. This is most common if your sampled audio emphasizes off-beats

You can shift all beat markers left or right by quarter beats by using the menu or key commands:

- From the **Sample Edit** menu, select either **Shift Beats 1/4 Beat Left** or **Shift Beats 1/4 Beat Right**.
- On your keyboard, press **Shift + →** to shift all beats right, or **Shift + ←** to shift all beats to the left.

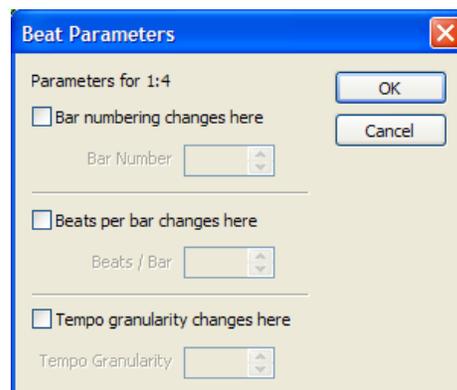
Beat Parameters

Using the Beat Parameter options, you can change the beat number, beats per bar, or tempo granularity.

To open the Beat Parameters dialog, right-click the beat where you want your change to begin.

When you open the Beat Parameters dialog, notice the affected bar and beat are listed in the dialog.

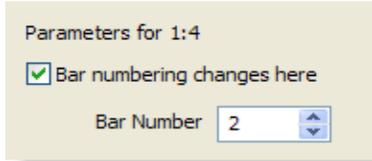
To keep track of a beat that has undergone manual changes, look for bar/beat indicators in boldface type.



Bar Numbering Changes

To change the bar number of a beat, check the box next to **Bar numbering changes here** here, and enter a new Bar Number. This command will make the current beat the first beat of the bar number you enter. Click **OK** to apply the changes.

Bar Numbering Changes may be necessary if an analyzed sample has a pick-up note leading to the first full bar. If the pick-up note gets labeled as 1:1, you might want to shift bar1 beyond that first note.



Beats Per Bar Changes

To change the number of beats per bar, check the box next to **Beats per bar changes here** here, and enter the number of beats per bar. This command changes the current bar and all subsequent bars to the length in beats that you enter. Click **OK** to apply the changes.



◆ **Note:** When you change the number of beats per bar, the currently selected bar becomes the first beat of that bar.

Tempo Granularity Changes

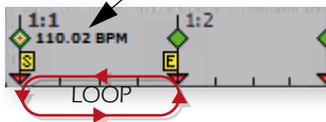
Tempo Granularity controls how often an original tempo is calculated. For example, a tempo granularity of 4 (the default setting for a 4-beat bar) means that an original tempo is calculated for every four beats. If you want the original tempo to be calculated on each beat, change the Tempo Granularity to 1. Lowering the Tempo Granularity can produce a tighter song, especially if you play multiple loops simultaneously. A higher Tempo Granularity, however, can produce a more realistic, less robotic sounding loop.

The tempo of a beat or bar also enables looped beats or bars to stay in synch with a tempo source. **In order to synchronize a beat or bar, that beat or bar must have a tempo assigned to it.** To assign a tempo to each beat, set the Tempo Granularity to 1.

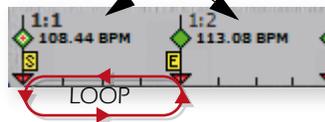
Look at the examples below. The first example does not have a tempo established for beat 1:2. This is because the tempo assigned to beat 1:1 is the tempo for a longer period of time than one beat.

The second example shows two beats, each with its own tempo. When you select a region between Beat 1 and Beat 2, it will stay in synchronized.

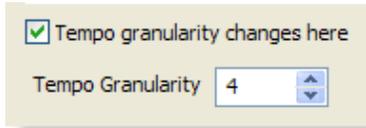
Will not Sync to Tempo because a tempo is not set for each beat.
(Tempo Granularity is greater than 1)



Will Sync to Tempo because a tempo is set for each beat.
(Tempo Granularity is 1)



To change the Tempo Granularity, check the box next to **Tempo granularity changes here** here, and set the granularity. This command sets the tempo granularity for the currently selected beat and all subsequent beats. Click **OK** to apply the changes.



Creating Loops from Your Analyzed Sample

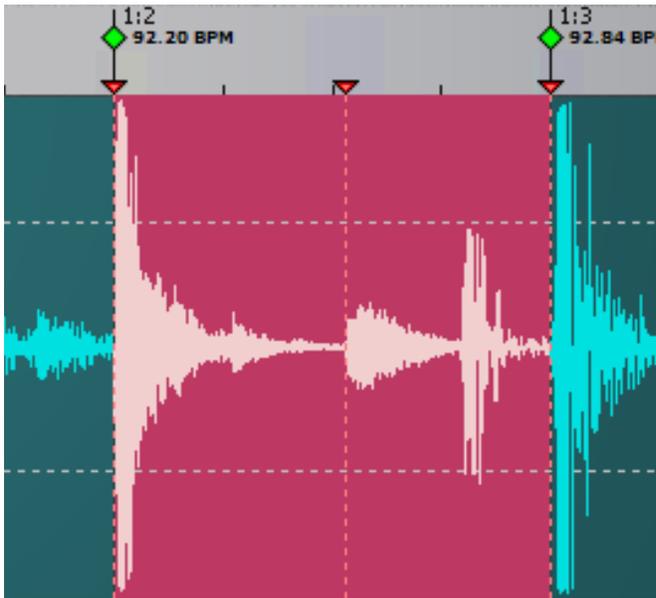
Once you've analyzed your sample, it's very easy to create loops within your sample. You can create as many loops as you want and assign them to different voices, assign continuous controllers to cycle through them, etc.

◆ For more information on creating loops, see ["Looping and Loop Processing" on page 203](#).

► To Create a Loop Within an Analyzed Sample:

1. Make sure **Snap to Marker**  is selected (this turns on by default when you run Beat Analysis).
2. Use your mouse to select the part of your sample you want to loop. For your loop to keep in sync with other loops, you must select an area between two Beat Markers that has a tempo.

■ **Tip:** Make sure that your loop begins and ends on a beat marker that has been assigned a bar and beat, and that the loop has a tempo.



3. From the **Sample Loop** menu, select **Make Loop From Selection** (or press Ctrl-Shift-L on your keyboard).
4. Click the **Loop** button to turn looping on (it should be on by default).



Tips for Making Selections for Loops

If you want to select the area between two beats for looping, there are some tricks that make it a little easier:

- Make sure **Snap to Marker**  is selected.
- You can show/hide Note Markers by clicking the  button in the toolbar, or by right-clicking the horizontal ruler, and selecting **Show Note Markers**.
- You can show/hide grid lines by clicking the  button in the toolbar, or by selecting **Show Grid** from the **Display** menu.
- If you want to select the length of a single beat (or a single measure if you're zoomed out), hold **Ctrl** on your keyboard while clicking a beat marker. The region between the beat you clicked and the next beat is selected. You can also right-click a beat marker and choose **Select** to accomplish the same thing.
- If you want to select an entire measure, zoom out until the individual beat markers disappear and you can only see the markers for first beat of each measure (1:1,2:1, etc.), then hold **Ctrl** on your keyboard while clicking the Beat Marker .

You can make as many loops as you want and use them for several purposes (which will be explained in further detail later). Loops can overlap, or include the entire sample.

If you make more than one loop within a sample:

- Select loops using the **Loop #** box above the sample display.  The currently selected loop displays yellow markers.
- You can view the markers associated with each loop by selecting the loop number from the toolbar.
- The loops are numbered in accordance to time, from left to right. This means Loop 1 always has the left-most start point. If you add a new loop before an existing loop, Loop numbers are re-aligned.
- To delete the currently selected loop, select **Delete Current Loop** from the **Sample Loop** menu.
- To delete all loops in the sample, select **Delete All Loops** from the **Sample Loop** menu.

◆ The currently selected loop appears with gray markers. The rest of your loops appear with yellow markers.

Using Loops

Once you create some loops in a sample, you can determine the way loops will behave when you assign that sample to a voice. For this, go to the **Voice Processing** page.

Playing Back Loops

Loop playback is very flexible, and you can assign continuous controllers, or even ranges of keys to determine which loop plays at any given time. When you assign each voice a sample containing one or more loops, by default, the loop behaves as follows:

- For each voice, you select which loop you want to repeat by using the **Loop (n)** knob in the Voice Processing window. The default loop is Loop 1.
- The **Start At Loop** button controls where playback begins. With **Start At Loop** turned on, playback begins at the beginning of the loop you selected using the **Loop (n)** knob. With it turned off, playback begins at the beginning of the sample.
- Once playback reaches the endpoint of the loop you selected in the Voice Processing window, it repeats that loop until you release the note.



■ **Tip:** If you want to trigger loops with your keyboard and have them continue playing without having to hold the note, set the voice to **Latch** in the Voice Processing window.

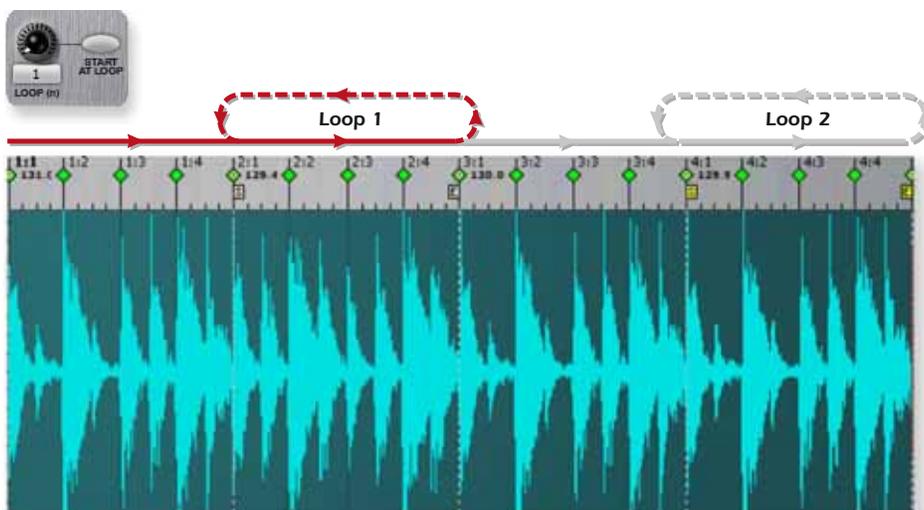
■ **Tip:** You can sustain loops by holding down a footswitch. When you want all sound to stop, release the footswitch.



Loop Select and Start At Loop

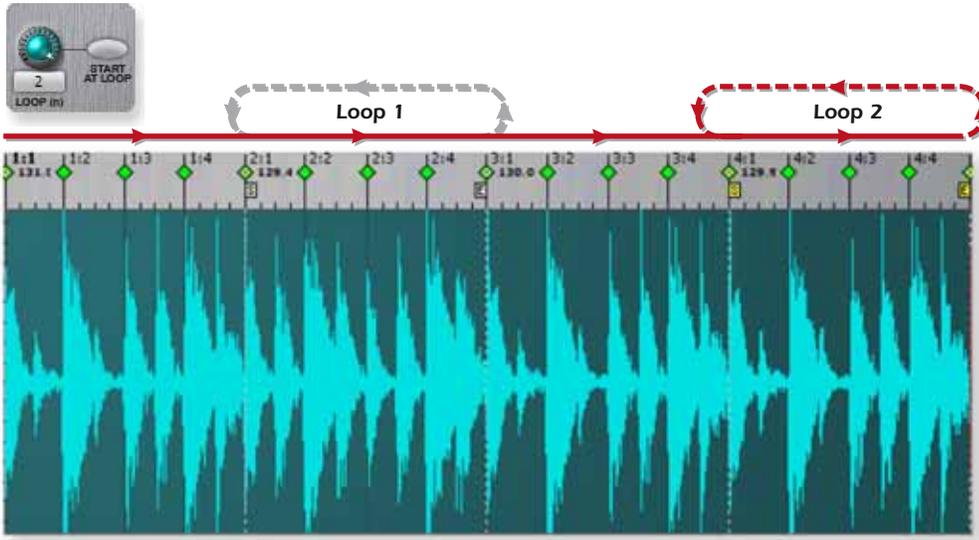
For example, let's say a sample has two loops, and you select Loop 1 in the Voice Processing window. When you play the note, the sample will begin at the beginning of the sample. When it reaches Loop 1's end point, it will jump back to Loop 1's start point. Loop 1 will repeat until you release the note.

Here is a visual representation:

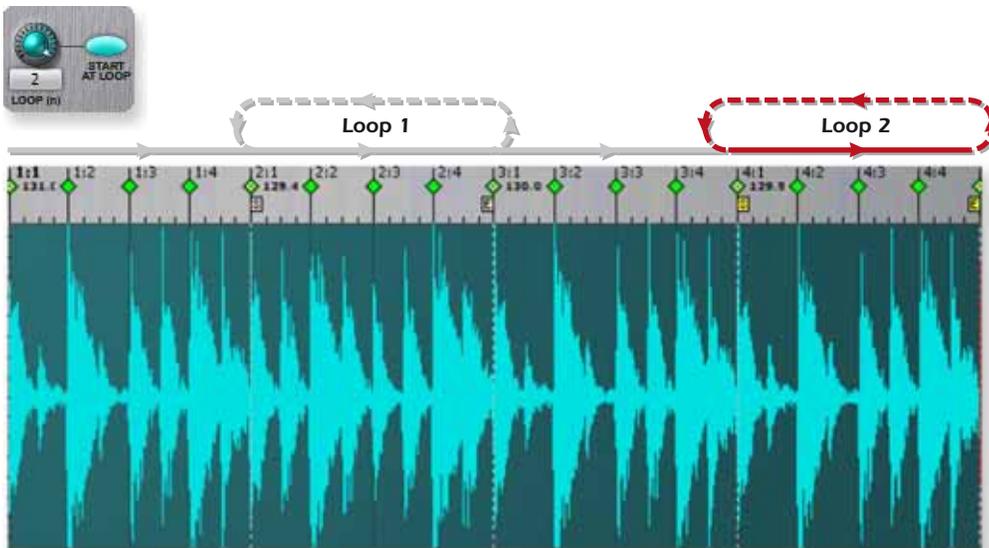


8 - TwistaLoop
Using Loops

Now, let's say you select Loop 2 on the **Loop (n)** knob in the Voice Processing window. Rather than repeating Loop 1, your sample plays until it reaches the end of Loop 2, and then it repeats Loop 2 until you release the note.



Now, take the above example, and select the **Start At Loop** button. Playback begins at the start point of Loop 2, rather than at the beginning of the sample:



Loop Speed and Tempo

The speed at which your loops play back is determined by whether you use the Twist-aLoop Override/Lock To Tempo ([page 73](#)), the Preset/TwistaLoop Lock to Tempo ([page 82](#)), or TwistaLoop Speed settings ([page 224](#)). With all these settings turned off, your loops will play back at their original tempo. With any of these setting turned on, the speed of the loops is altered. However, the pitch does not change.

▼ **Note:** You cannot use the **Chorus** settings on voices that have time-stretching enabled. This includes syncing with tempo and loop speed changes.

Changing Loops Using a Continuous Controller

There are two Cord destinations available for controlling which loop repeats:

- Loop Select (Continuous)
- Loop Select (Jump)

You can set up any source for these cords, but you'll probably find a continuous controller to be the most useful.

When using the Loop Select cords, you should bear the following in mind:

- The **Start At Loop** button determines whether playback begins from the beginning of the sample, or from the start point of the loop.
- You should keep the **Loop (n)** knob set to 1 (or 0% in the case of multisamples). This allows you access to all your loops. Use the Cord's Source to control the initial loop.



You can do this by either using the source to determine the initial loop (e.g. moving your continuous controller) before you play a note, or by setting up the source's Initial Controller amounts in the Preset window. For more information on Initial Controller Amounts, see ["Initial Controllers A-P" on page 81](#).

Loop Select (Continuous)

If you want to change which loop repeats while playing a note, without skipping any of the sample between the loops, you can set up the **Loop Select (Continuous)** cord to do so. This way, when playing back the sample, changing the CC value causes playback to move to the selected loop.

This example assumes your sample contains two loops. As you experiment with TwistaLoop™, you're best off using distinct sounding loops with some space between them to be able to distinguish which loop is playing.

► To Change Loops Using a Continuous Controller

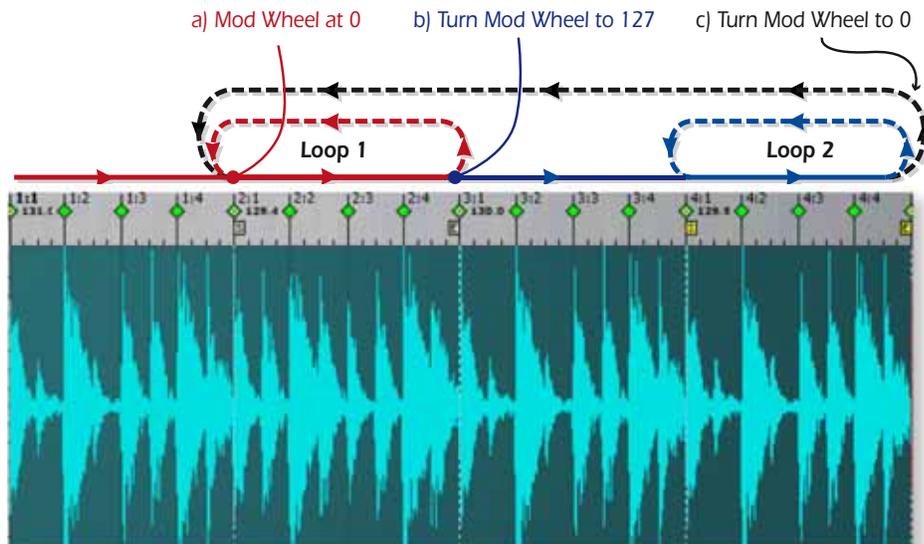
1. In the Voice Processing window, select the group containing the voice you want to control.
2. Set the **Loop (n)** to 1. De-select **Start At Loop**.
3. Find an unused Cord, and set the Source to the continuous controller you want to use (e.g. **Mod Wheel**, **MIDI A**, etc.).
4. Set the Cord Amount to **+100**.
5. Set the Destination as **Loop Select (Continuous)**. Your Cord should look like this:



- If Loop 1 doesn't fall at the very beginning of the sample, and you want playback to begin at the start point of Loop 1 (rather than at the beginning of the sample), select **Start At Loop**.

Let's say you chose MIDI A as the source. Now, when you play back the sample, looping behaves as follows:

- a) If you don't touch the MIDI A continuous controller, playback begins *at the beginning of the sample*. Once you reach the endpoint of Loop 1, Loop 1 repeats until you move MIDI A, or release the note.
- b) While still playing the note, if you move MIDI A to 127, Loop 1 continues playing until it reaches its endpoint. Then, *the rest of the sample plays until you reach the endpoint of Loop 2*. Loop 2 then repeats.
- c) While still playing the note, if you move MIDI A back to 0, Loop 2 continues playing until it reaches its endpoint. Then, *playback jumps to the startpoint of Loop 1*.



Loop Select (Jump)

If you want to jump from one loop to another without releasing a note, you can set up the Loop Select (Jump) Cord to do so. This way, changing a CC value causes playback to jump to the newly selected loop once it reaches the endpoint of the current loop.

► To Jump Loops Using a Continuous Controller

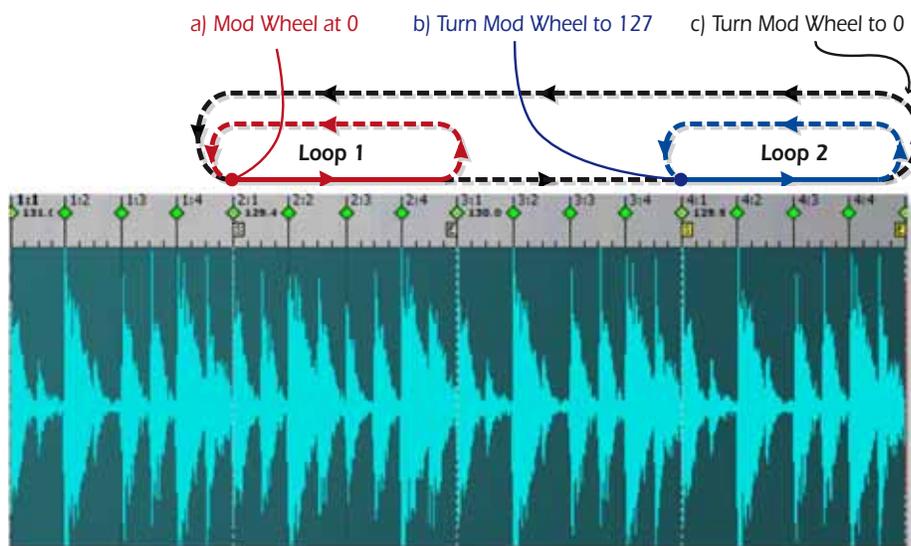
1. In the Voice Processing window, select the group containing the voice you want to control.
2. Set Loop (n) to 1. This time try selecting **Start At Loop**.
3. Find an unused Cord, and set the Source to the continuous controller you want to use (e.g. Mod Wheel, MIDI A, etc.).
4. Set the Cord Amount to +100.
5. Set the Destination as **Loop Select (Jump)**. Your Cord should look like this:



■ If Loop 1 doesn't fall at the very beginning of the sample, and you want the beginning of the sample to play first, don't select **START AT LOOP**. This is useful for pickup notes, or introductions.

Let's say you chose **MIDI A** as the source. Now, when you play back the sample, looping behaves as follows:

- a) If you don't touch the MIDI A controller, *playback begins at the startpoint of Loop 1*. Once you reach the end of Loop 1, Loop 1 repeats until you move the MIDI A, or release the note.
- b) While still playing the note, if you move MIDI A to 127, Loop 1 continues playing until it reaches its endpoint. *Playback jumps to the startpoint of Loop 2*. Loop 2 then repeats.
- c) While still playing the note, if you move MIDI A back to 0, Loop 2 continues playing until it reaches its endpoint. Then, *playback jumps to the startpoint of Loop 1*.



Starting at a loop other than Loop 1 - Unless you have **Start At Loop** selected, if you use MIDI A to set an initial loop other than Loop 1, you *cannot* jump to a different loop until the current loop finishes. This means that your sample must play from the beginning of the sample to the end of the loop that you selected as the initial loop before jumping to another loop.

Other Ways to Change Loops

To change loops you can use any Cord source, such as **Velocity** or **Pedal**, in conjunction with the Loop Select Cords. One particularly useful source is the **Key CC Win Cord**.

The **Key CC Win** cord allows you to set up a range of keys on your controller to control a Cord destination. Here's an example.

► To Select Loops Using the Key CC Win Cord

1. In the Voice Processing window, select the group containing the voice you want to control.
2. Set the **Loop (n)** to 1. Select **Start At Loop** if you want to start at the beginning of the selected loop.
3. Find an unused Cord, and set the Source to **Key CC Win**.
4. Set the Cord Amount to **+100**.
5. Set the Destination to **Loop Select (Continuous)** or **Loop Select (Jump)**. Your Cord should look something like this:



6. In the tree to the left, go to the Preset Screen by clicking the icon with the little keyboard  icon for the current preset.
7. In the **Key CC Window** area, click the **On** button to illuminate it in blue.
8. Select **Low** and **High** values. These will be the keys on your controller that you'll use to select loops. Since they will no longer play any notes, you should select a range of keys that you don't anticipate using for this preset.



Now, when you play back your sample, the notes on your keyboard that you assigned to the Key CC Window range will control the current loop. Try it! Play the looped voice on your keyboard. While still holding the note, change loops using one of your Key CC Window keys and hear the loop change.

■ **Tip:** Select a Key CC Win range that has the same number of notes, including black keys, as loops you've created in your sample. For example, if your sample has five loops, use a range of five notes, such as C0 to E0.

Synchronizing Loops

Loops can synchronize to an internal clock source, a MIDI Clock, or a VST Host clock. For more information on clock sources, see ["Tempo Control" on page 74](#).

If you want beats to synchronize to a tempo:

- Make sure either **Lock Oscillators To Tempo** is selected in the Multisetup window, or that the preset is locked to a tempo channel. This will make your loops synch to whatever source you are using as a clock.
- Loops can be twisted (e.g. with TwistaLoop speed, or a tempo source) between 50 and 200 percent of their original tempo.
- Loop with Start and End points must lie on beat/bar markers.
- Loops must have a tempo. This means that if you're looping an individual beat, that beat **MUST** have a tempo assigned specifically to it. If you want a beat to have a tempo, change the Tempo Granularity. [See "Tempo Granularity Changes" on page 214](#).

Multisample Voices and Looping

You might wonder what happens if you want to use a multisample voice that contains samples with multiple loops. How do you select the initial loop if each sample contains a different number of loops?

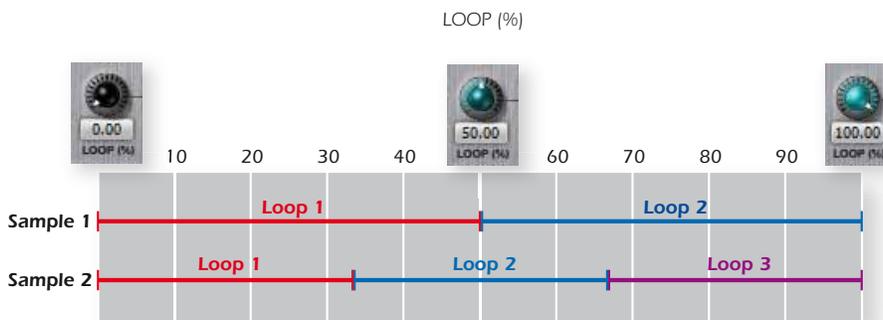
When you use a multisample voice, the **Loop (n)** knob becomes the **Loop (%)** knob. When you create multiple loops in each sample, the knob is divided into percentages based upon the number of loops in each sample.

For example, if one sample contains two loops, a setting of 0% to 49.99% selects the first sample. A setting of 50% to 100% selects the second sample.

If another sample contains three loops, a setting of 0% to 33.33% selects the first sample, 33.34% to 66.66% selects the second, and 66.67% to 100% selects the third.

So, if you use both these samples in a multisample voice, you'll see why this setting is necessary. In the above example, with the two samples combined, the knob position determines which sample to select for both samples.

This diagram illustrates the concept:



Notice that a setting of 30% selects Loop 1 from both samples in the multisample voice. However, 40% selects Loop 1 from the first sample, and Loop 2 from the second.

The Speed Knob

The TwistaLoop Speed knob in the Voice Processing window serves a special function. Unlike the Tempo knob, which controls the tempo of Emulator X, the Speed knob controls the speed of any loops for *individual* voices and groups.

Changing a Loop's Speed

Use the Speed knob if you simply want to control the speed of loops on an individual basis. The Speed knob is most useful if you're not trying to mix loops together in sync with each other,

► To Change a Loop's Speed

1. For the current preset, click **Voice Processing** in the tree on the left.
2. From the **Group** drop-down menu, select the group that contains the voices you want to alter.
3. Turn the TwistaLoop™ Speed to the desired percentage. The loop's original tempo is 100%. Reducing the percentage slows down the loop, and increasing the percentage speeds it up.



▼ **Note:** For loops that are synched to a clock source, you are best off leaving the **Speed** set to **Off**, unless you want to be able to nudge the tempo.



Nudging the Tempo

Nudging the tempo is particularly useful if you are using the multisetup and want to crossfade between two presets on different MIDI channels and keep them in sync with each other, and with your master tempo.

► To Nudge the Tempo With the Pitch Wheel

1. For the current preset, click **Voice Processing** in the tree on the left.
2. From the **Group** drop-down menu, select the group you want to be able to nudge. If you're using a Multisetup, you'll probably want to select **All**.
3. For the **Bend** settings, change both **Up** and **Down** to 0.



4. Set the TwistaLoop Speed to 100%.



5. Find the Cord that uses **Pitch Wheel** as its source (Cord 2 by default).
6. Change the **Destination** to **Osc Speed**, and the **Amount** to **10**. Your Cord should look like this:



Now, while you play back the preset, you can use the pitch wheel to change the tempo. Since you're controlling the tempo with your pitch wheel, the tempo always snaps back to 100% when you let the pitch wheel go. This allows you to nudge the tempo far enough to synch it up with loops playing on other voices or channels.

Using Regions of Analyzed Samples

Another way you can use a sample that contains bar, beat, and loop information, is with the Regions page in the Voices and Zones window. You can use the same sample for multiple voices, and select different beats or bars for each voice. This allows you to map different beats or bars from a single sample to different keys.

As with loops defined in the Sample Edit window, you can lock your region-based forward loops to a clock source and modulate them using Tempo knobs.

To define regions, first perform Beat Analysis on a sample. Then, using this sample, create a new preset and go to the Voices and Zones window.

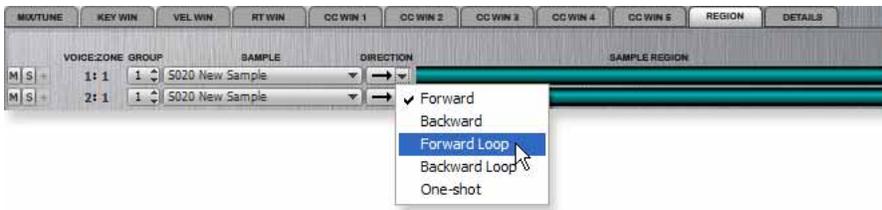
For specific instructions on using the Region page, see [“Region Page” on page 115](#).

► To Set Up Multiple Voices Using the Same Sample

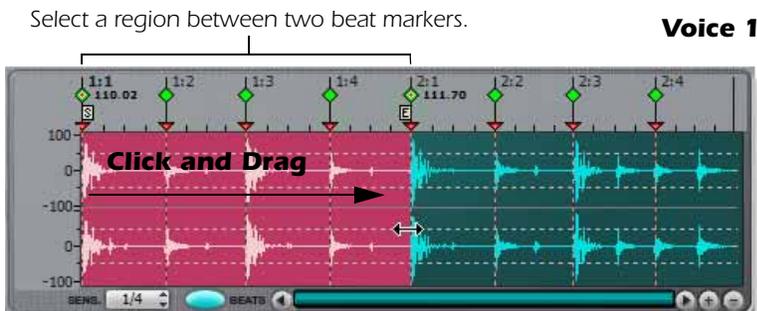
1. Make a new preset by clicking the Presets folder in the Sampler tree to the left, and selecting **New Preset** from the Preset menu.
2. In the new preset, select **Voices and Zones**. There should be one voice already there.
3. Click the **Region** tab.
4. In the sampler tree to the left, expand the **Samples** folder (but don't click any samples).
5. Click and hold a sample you've analyzed. While still holding down the mouse button, drag the sample to C3 on the keyboard at the bottom of the Voices and Zones window. This creates a new voice, with the sample you selected mapped to C3.
6. Repeat step 4, this time dragging the same sample to D3.
7. Delete the empty voice that was initially in the preset by right-clicking the voice and selecting **Voice, Delete Voice**.

► To Use Different Loops For Each Voice

8. For each voice, select **Forward Loop** from the **Direction** drop-down menu.

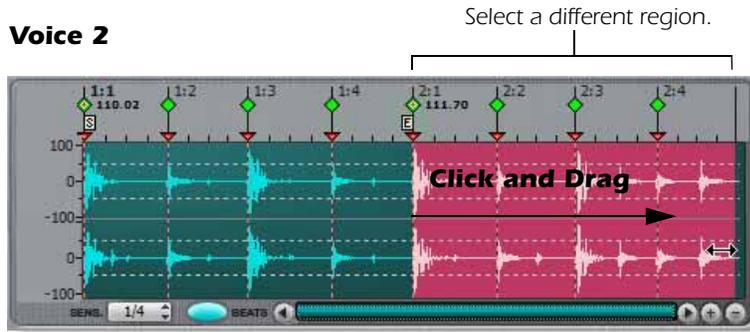


9. Select **Voice 1** by clicking it.
10. In the waveform display at the bottom of the window, use your mouse to select a portion of the sample. Select a region between two beat markers.



◆ To select a region between beat markers, make sure the ruler format is **Bars:Beats**, and that **Show Beat Markers** is selected.

11. Select the other voice and repeat step 9, only selecting a different region of the sample.



■ **Tip:** To easily select an area between two beat markers, you can hide Note Markers by clicking the  button in the toolbar.

Now, when you play C3 on your controller, the region selected in Voice 1 loops. When you play D1 on your controller, the region selected in Voice 2 loops.

Play around, try making more voices and mapping different regions to them. You are not limited to selecting by beats; you can select an area between Note Markers or Loop points. Or, if you hide all markers, you can freely select any region of the sample.

However, region-based loops will only lock to the master or preset tempo if **your regions are defined between two beat markers AND set to forward looping.**

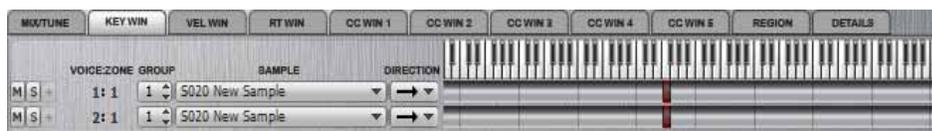
◆ **Note:** You cannot lock backward-looped regions to a clock source.

Using the RT Win With Regions

You can do some creative things with real-time crossfades and regions. For example, if you use a beat-analyzed sample, such as a drum beat, with multiple voices, you can set each voice to use a different region of the same sample. Then you can switch and crossfade between the two regions in real time. Using beat markers to define the regions, in conjunction with the Lock To Tempo feature ensures that all the voices stay in sync with each other.

► To Set up The RT Win to Crossfade Between Two Loops

1. Set up several voices, each using a different region of the same sample with forward looping.
2. Make sure that the voices are mapped to *the same keyrange* on your keyboard.



Map both voices to the same keyrange.

3. Use the RT Win page to set up a crossfade between the voices.



4. Make sure that **Lock To Tempo** is turned on in the Multisetup or Preset window.
5. In the Voice Processing window, set up a Cord for both voices that controls **Amp Crossfade** with a continuous controller.



Now, you can use the continuous controller during playback to crossfade between two forward-looped regions. Since the crossfade works in real-time, you don't need to re-trigger the note-- just change the continuous controller with your finger still on the note. If you select Lock to Tempo in the Multisetup window, the voices will stay in sync with each other. For more information, see ["Tempo Control" on page 74](#).

If you choose to lock looped regions to a tempo source, you **MUST** select a region with Start and End points that lie on beat/bar markers. In addition, the beat or bar **MUST** have a tempo. If you want a beat to have a tempo, change the Tempo Granularity. [see Tempo Granularity Changes on page 214](#).

■ **Tip:** If you want to sustain a looping voice without having to hold down the key, turn on **Latch** in the Voice Processing window. Make sure to select the right group from the **Group** drop-down menu.

Troubleshooting

Having trouble getting TwistaLoop to do what you want? Check out these troubleshooting tips.

Beat Analysis Doesn't Find the Right Beats

You analyzed a sample using Beat Analysis, but the Beat Markers do not fall in the correct places.

- In the Beat Analysis dialog, make sure you use an appropriate tempo range. Try to estimate the tempo as best you can. Also, make sure you select the correct Beats per Bar and Meter.
- In the Beat Analysis dialog, experiment with the Tempo Variation. If your sample changes tempo frequently, try using **Fast**. If the tempo is tight, and unchanging, try **Constant**.
- Use your ears and remove any extra beats from the beginning and/or end of the sample. To delete part of a sample, select the area you wish to truncate, and press the **Del** key on your keyboard. Now, re-analyze the sample.

Looping Doesn't Occur

You've defined a loop's start and endpoints and assigned the sample to a voice. Yet, the defined loop doesn't seem to repeat.

- For the sample you're trying to loop, make sure that the Loop button  is turned on in the Sample Edit window.
- Make sure you're actually getting to the looped area during playback. Remember, if you have the **Start At Loop** button de-selected, playback begins at the start of the sample and doesn't repeat the loop until it reaches the current loop's end marker.
- Make sure you haven't inadvertently selected **Forward Loop** in the Voices and Zones window. You cannot use both methods of looping simultaneously. If you want to use TwistaLoop with Loop Start and Loop End markers, all the voices that use that sample must have their direction set to **Forward**.

Loops Don't Synchronize to Tempo

You have one or more loops and you want them to stay in time with your clock source, but they do not modulate correctly, or they drift out of time.

- Make sure that your loop Start and End points occur on Beat Markers .
- Loops can be twisted (e.g. with TwistaLoop speed, or a tempo source) between 50 and 200 percent of their original tempo. Make sure you aren't exceeding this amount.
- If you're looping regions, make sure that the regions begin and end on Beat Markers.
- Make sure you have the proper Tempo Granularity set. If you're looping one beat, the Tempo Granularity must be set to 1. [see Tempo Granularity Changes on page 214.](#)
- Make sure that **Lock To Tempo** is turned on in either the Preset Globals window, or in the Multisetup.
- More things to check:
 1. Click the **MIDI** tab in the Preferences dialog.
 2. Make sure you have selected the correct Tempo Source information. If you want to lock to the Emulator X clock, or to a VST host, make sure **External Tempo Source** is unchecked.
- If you're using the preset to control tempo, make sure that you select a Preset Tempo Channel. For more information on the Preset Tempo Channel, see ["Preset Tempo Channel" on page 27.](#)

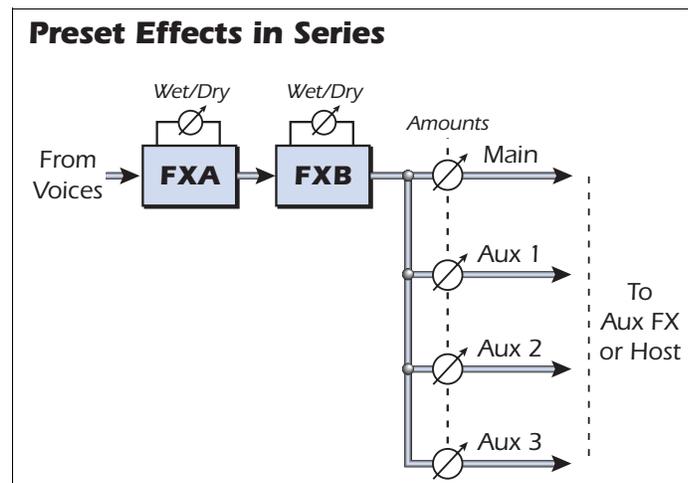
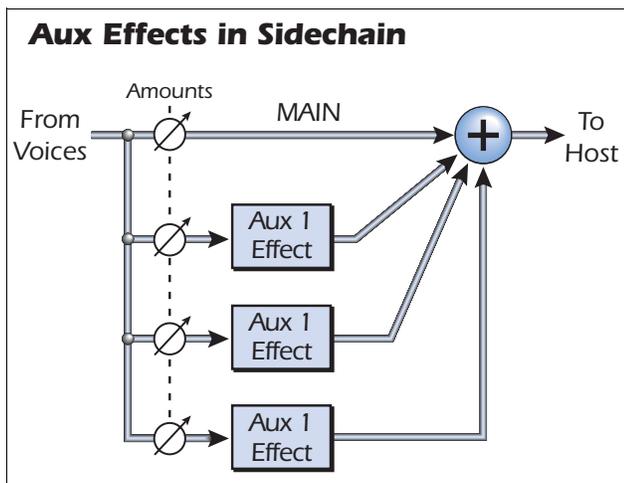
9 - Effects

Overview

In addition to the hardware-based effects of PatchMix DSP, Emulator X offers software effects processors and a powerful effects routing scheme. The E-MU engineers have created a complete collection of top quality effects using 32-bit floating point precision for pristine sound quality. In addition, the effects algorithms have been designed for low CPU usage.

The Emulator X effect routing architecture is extremely flexible. Effects can be included as part of the preset or multisetup or both. Preset FX and Aux FX both use the same effects algorithms.

- **Multisetup (Aux) Effects** are designed to be used for your main effects such as reverb or delay. Aux effects can be applied to any or all of the 64 MIDI channels.
- **Preset Effects** are used when you want an effect to be included as part of a certain preset. For example, you might want to add chorus to a string pad or distortion to a guitar sound.



Each effect has a number of user-adjustable parameters and a wet/dry mix control. See [page 28](#) to learn how to connect controllers.

When using effects as part of the preset, the wet/dry mix or effect amount is controllable on a per-voice basis. This means that you can create presets using effects as integral parts of the sound and control the effect amounts using LFOs, envelope generators, key velocity or any other PatchCord source. As an example, you could build a preset where the distortion amount increases the harder you hit the key.

Programming Effects

Adding an Aux Effect in the Multisetup

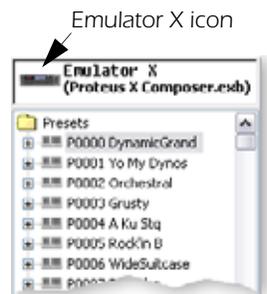
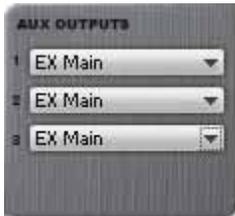
Some banks (Proteus X Composer, for example) are already programmed to use Emulator X soft effects. These instructions explain how to add Aux Effects to an Emulator X bank which does not already have soft effects programmed.

The Aux Effects are designed to be used as sidechain effects applied on all presets in the bank. See the diagram on [page 234](#) for a complete description of the effects routing.

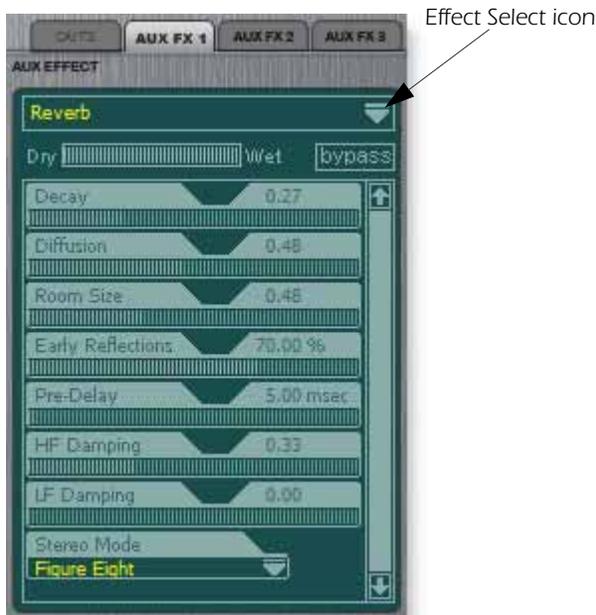
- PatchMix DSP hardware effects routings are already programmed into most Emulator X banks

► How to Add an Aux Effect

1. Select the Emulator X Studio session in PatchMix DSP.
2. Close PatchMix DSP
3. Select the Multisetup Page by clicking the **Emulator X icon** at the top of the tree.
4. Locate the **Aux Outputs** section in the lower right part of the window and set all the Aux Outputs to EX Main (or the main output of Emulator X). This bypasses the PatchMix DSP hardware effects and sets up the Aux Outs to be used as sidechain effects.



5. Click on the **Aux 1** button at the top right of the screen if it's not already selected. The Aux 1 effect appears in the TV screen.
6. Click on the **Effect Select Icon** at the top right corner of the Effects TV screen. A list of available effects drops down.

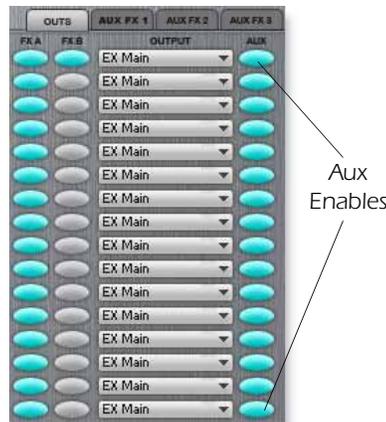


7. Select one of the Effects from the list.
8. Turn up the FXA knob in the MIDI controllers section to hear the effect. (FXA is usually located on Controller N.)
9. **Change the effect parameters** to suit your own taste, or change the entire effect for that matter. Keep in mind that the changes you make will affect **ALL** the presets in the bank.
10. **Save the bank** if you want to keep your changes.

► **How to Temporarily Disable the Aux Effects**

You might want to turn off the Aux effects to free up CPU or just to hear the dry mix.

1. Select the **Multisetup Page** by clicking the Emulator X icon at the top of the tree.
2. Click on the **OUTS** button near the top right part of the screen. The screen shown at the right appears.
3. The row of buttons marked **AUX** disable the Aux outputs for each channel.
4. **Turn the buttons off (grey)** for the channels on which you don't want effects.



◆ Emulator X factory banks have the FXA, FXB and FXC controllers already programmed. When creating your own presets and banks, we recommend you follow the FX routing model on [page 237](#).

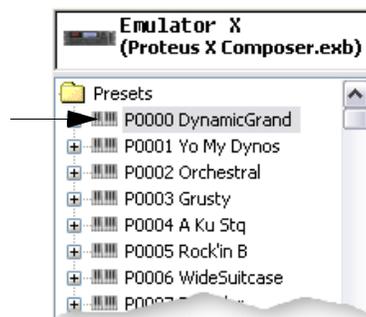
◆ The Aux buttons turn all three Aux busses On or Off.

Adding an Effect to a Preset

Suppose you have a nice string sound, but you want to make it really thick and lush. The obvious solution is to add a chorus. Since you only want the chorus to affect the string sound, you'll use a preset effect.

► **How to Add a Preset Effect**

1. Select the preset you want in the tree by clicking on the keyboard icon. The Preset Global page appears.
2. Click on the **Effect Select Icon** at the top right corner of the Effects TV screen. A list of available effects drops down.
3. Select **Chorus** (or whatever effect you wish) from the list. The Effect name is now listed above the TV screen and the effect parameters are shown in the TV screen.
4. **Play the keyboard.** You should now hear the selected effect as part of your sound. If you don't hear effects, make sure the FXA & FXB enable buttons are on for the MIDI channel you are using. [See page 234](#).
5. Adjust the **Wet/Dry mix** to control the effect amount.
6. Adjust the effect parameters until you find the sound you want or select a Template. [See page 279](#).
7. **Save the bank** if you want to keep your changes.

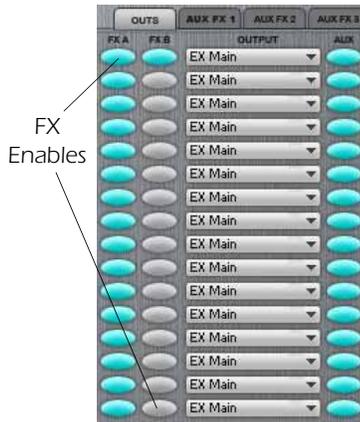


▼ Remember that if a preset with effects is used on more than one MIDI channel, the CPU penalty multiplies each time the preset is used.

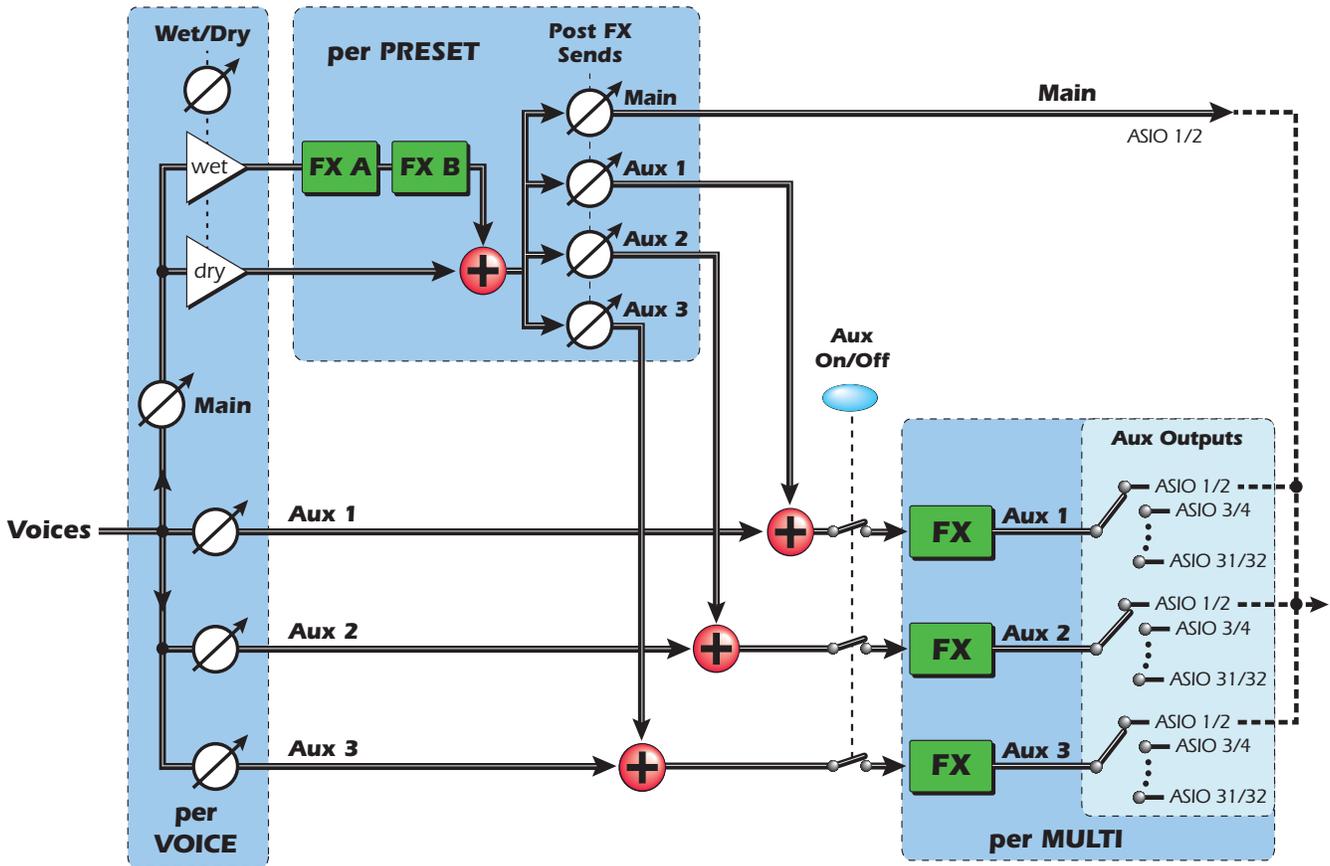
► **How to Temporarily Disable a Preset Effect**

1. Select the Multisetup Page by clicking the Emulator X icon at the top of the tree.
2. Click on the OUTS button on the top left of the screen. The screen shown at the right appears:
3. The two rows of buttons marked FXA and FXB enable or disable the preset effects for each channel.

Turn the buttons off (grey) for the channels you are using if you don't want to use the effects.



Block Diagram of the Effects Routing

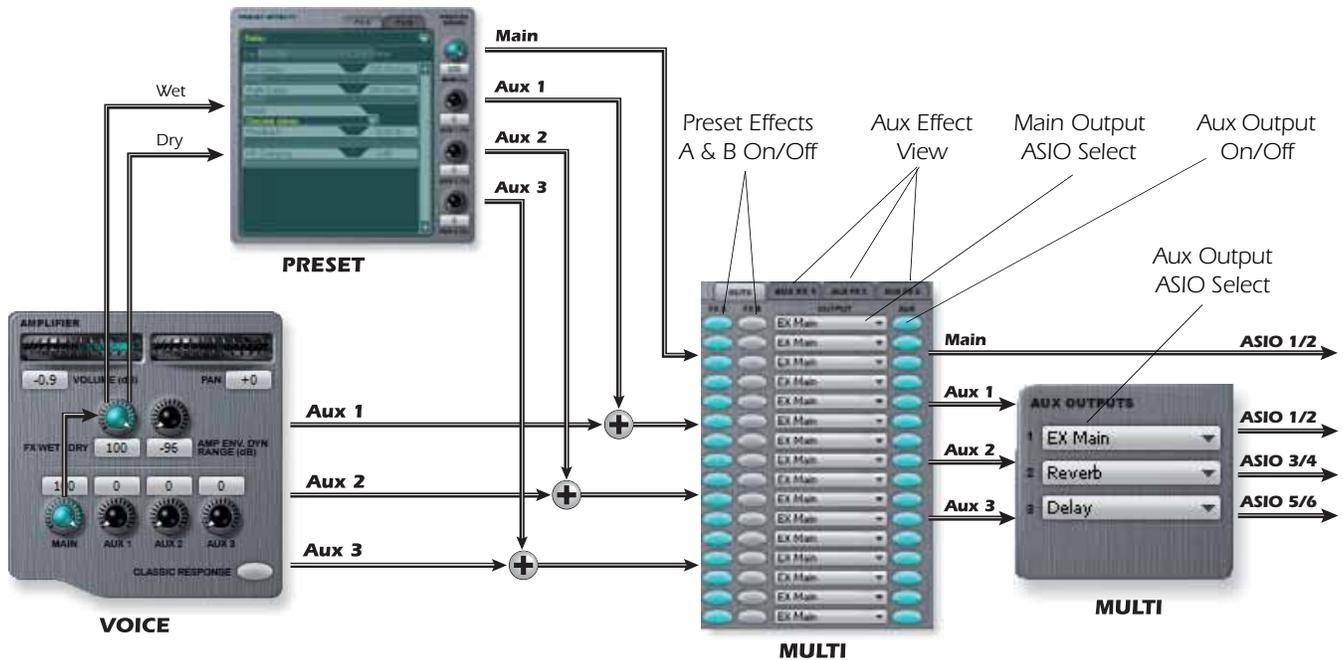


Two effects can be assigned per Preset. Three effects can be assigned per Multisetup. Effects amounts can be controlled at the voice level or the preset level. Note that since the Main output and Aux 1, 2 & 3 are all connected to the same ASIO output, they will be MIXED before being sent to the Host application or soundcard.

FX Routing

Make sure that you are thoroughly familiar with the architecture of Emulator X ([page 31](#)), before delving into the effects section and it will make a lot more sense. Study the effect routing diagrams below and the preceding page. The effects architecture of Emulator X is very flexible and allows for a variety of different signal routing schemes.

Block Diagram of the Effects Screens



This diagram shows the effects routing using the actual Emulator X screens in the Voice, Preset and Multisetup. The Main output and the Aux 1 outputs are both routed to EX Main (ASIO 1/2) and will be combined into the ASIO 1/2 channels.

Voice - Main Output	Adjusts the amount of Voice signal to the Wet/Dry mix control.
Voice - Aux Outputs 1-3	Adjusts the amount of Voice signal sent to the Aux 1-3 Busses (and Effects)
Voice - Wet/Dry Mix	Adjusts the Wet/Dry mix of the Voice main output sent to the Preset effects section.
Preset - Main Output	Adjusts the amount of Preset effect signal that will be sent to the Main Output.
Preset - Aux Outputs 1-3	Adjusts the amount of Preset effect signal that will be sent to the Aux 1-3 busses.
Multi - Effects A & B On/Off	These buttons enable or disable the Preset A & B effects for each MIDI channel.
Multi - Aux Effect View	These buttons let you see the Main output screen or the Aux 1-3 effect screens.
Multi - Main Output ASIO	This field allows you to select the ASIO channel pair for the Main Output. ASIO outputs appear in the list only after they have been created in PatchMix DSP.
Multi - Aux Output On/Off	This button turns the Aux outputs on or off for each MIDI channel. In the factory presets, this button effectively enables or disables the global effects.
Multi - Aux Output ASIO	This field allows you to select the ASIO channel pair for Aux Outputs 1-3. ASIO outputs appear in the list only after they have been created in PatchMix DSP.

You can place effects in three places:

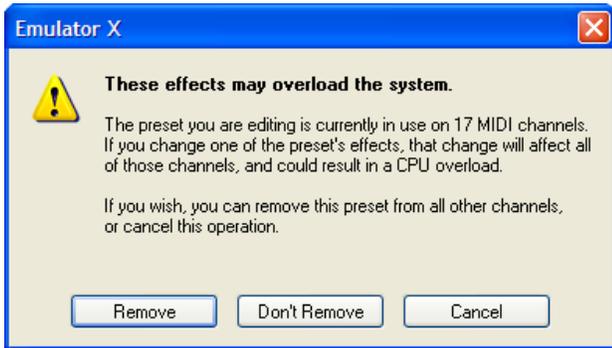
- **Multisetup** - - - - - Effects placed in the Multisetup (Aux 1, Aux 2 or Aux 3) can be used by all Presets in the Bank.
- **Preset** - - - - - Effects placed in the Preset (FXA or FXB) are used by one specific Preset.
- **PatchMix DSP** - - - - - Hardware effects can be applied in the PatchMix DSP mixer application. These effects use minimal CPU resources, but do not get saved with the Bank. You have to call up the appropriate PatchMix Session.

▼ If a Preset containing an effect is used by more than one MIDI channel, more CPU resources are used.

Effects Overhead

Each effect you select, whether it is located in a Preset or in the Multisetup, uses a percentage of your computer's CPU. Extensive use of effects in every preset can easily bog down your computer. Effects begin running (and using CPU) as soon as they are selected.

The following dialog box appears if you try to add an effect to a preset which is used on multiple MIDI channels.



Typical Effects Usage

Because the software effects use precious CPU resources, you will generally use Aux effects in the Multisetup for your main effects, since these can be shared by all presets in the bank. You can still use Preset Effects as needed, keeping in mind that the more software effects you use, the fewer CPU resources will be available for polyphony, recording and other tasks.

ASIO Outputs

ASIO outputs appear in the list only after they have been created in PatchMix DSP. If you only have three ASIO input strips in PatchMix DSP, you will only see three choices in the Multisetup ASIO Selects. The ASIO Outputs take their names from the names of the Strips in PatchMix DSP.

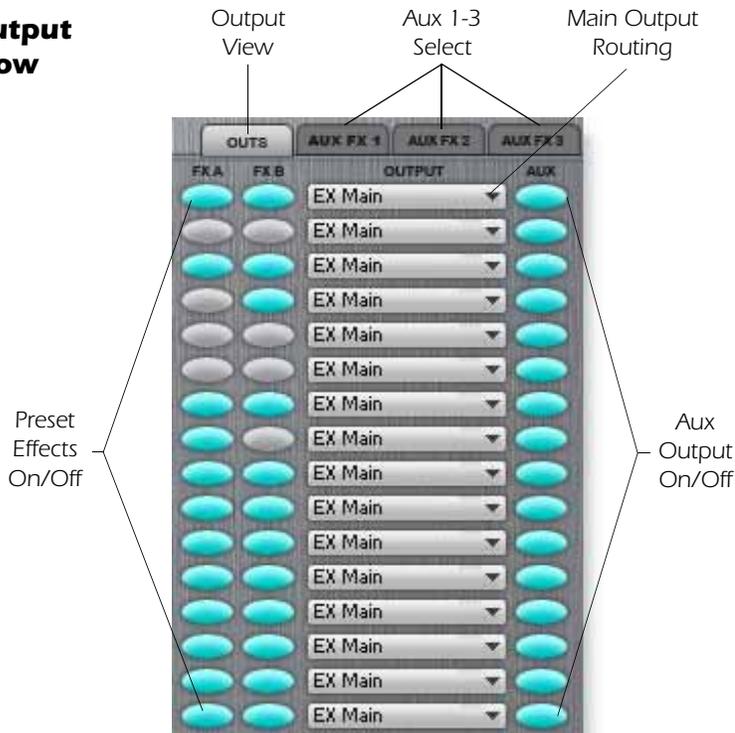
Main Output Window

The Main Output window is accessed by clicking the **Outs** button. (This option is not available in "Single" mode.) This one window allows you to set the ASIO routing of the Main Output, enable the Preset Effects, and enable the Aux Outputs for each MIDI channel. This window is extremely handy for turning all your effects on or off.

The **Preset Effect On/Off** buttons are used to temporarily disable the selected A or B effect while passing the signal. These controls allow you to hear each preset with or without effects or to turn off all Preset Effects in order to gauge the CPU load incurred by the Preset Effects.

The **Aux Output On/Off** buttons work differently than the Preset Effect On/Off buttons and *completely* turn off the three Aux Sends for the selected channel. When using the factory presets, the Aux Output buttons are used to enable or disable the Aux Effects (because all outputs are connected to EX Main). If you were using the Aux busses to send signals to physical outputs, the Aux button could be used to turn these outputs on or off.

Main Output Window



Output View	Selects the Main Output Window. (Note: This button is disabled in "Single" mode.)
Aux 1-3 Select	Select between Aux 1, Aux 2, or Aux 3 FX in the TV view.
Preset Effects On/Off	Turns the Preset A or B effects On or Off for each MIDI channel and passes the signal through.
Main Output Routing	Selects the ASIO channel for the Main Output. See tip ->
Aux Output On/Off	Turns the Aux Outputs On or Off for the selected channel. This turns all three Aux Outputs Off, rather than bypassing the effects. (This button functions as an FX Bypass when the standard Proteus X Composer template is used.)

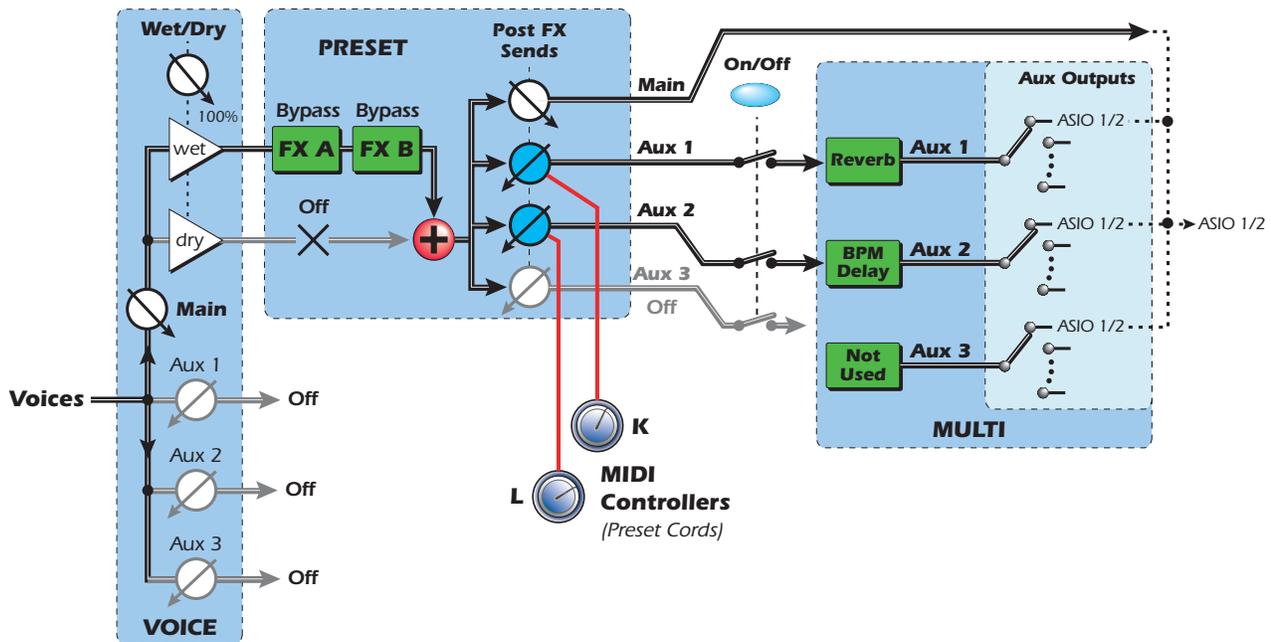
■ ASIO channels must be created in PatchMix or another Host application before they appear in this list.

Proteus X Composer FX Template

This is the effects architecture we have chosen to use for the Proteus X Composer bank. We recommend that you use this scheme as a model for your own presets unless you have special requirements. Preset effects and global effects can be used simultaneously and you can use PatchMix DSP hardware effects by simply changing the Aux Output routing.

The main effects are located in the **Multisetup** so that they can be shared by all presets. This is the standard sidechain effects routing scheme found on all mixing boards. **Aux 1 = Reverb** and **Aux 2 = BPM Delay**. MIDI controller K controls Reverb amount and MIDI controller L controls Delay amount through the Preset PatchCords.

The diagram below shows the Proteus X Composer routing scheme.



Key Points

Voice

- Aux 1, 2, 3 are turned fully down and are not used.
- The Main Output is turned up full.
- The Wet/Dry Mix is turned to full wet.

Preset

- No effects are selected for Preset FXA/FXB. (To add a preset effect, just select one!)
- The Main Output is turned up full.
- Aux 1 and Aux 2 are initially turned down, but they are controlled by Preset PatchCords connected to MIDI controllers K & L.
- Aux 3 is turned down and is not used.

Multisetup

- Aux 1 is routed to Reverb.
- Aux 2 is routed to BPM Delay.
- **Important:** All the Aux Outputs are set to ASIO 1/2 and so they are mixed together before being sent to the Host computer.

Controlling the Preset Effect Amount from the Voice

Emulator X allows you to control the Preset effects amount from the voice. This means that the amount is individually controllable for each key press. With this great feature you can create incredible effects such as:

- Velocity or pressure controlled distortion on guitars, organs or electric pianos
- Release velocity controlled echoes or reverb
- Chorus or Flanger amount controlled by an envelope generator or LFO.
- Ring Modulator or Pitch Shifter effects that only occur during the attack.

Because the effect amounts are now controlled by your performance, this technique contains endless possibilities. Try it!

► How to Control the Effect Amount from the Voice

For this example, let's set up the keyboard so that echoes only occur with low velocity. When you hit the keys soft, you get echoes; when you play hard, you get no echoes.

1. Select the preset you want in the tree by clicking on the keyboard  icon. The Preset Global page appears. (Choose a sound with a fast release for this example.)
2. Click on the Effect Select icon  at the top right corner of the Effects TV screen. A list of available effects drops down.
3. Select Delay from the list. The Effect name is now listed above the TV screen and the effect parameters are shown in the TV screen.
4. Play the keyboard. You should now hear the echo. Adjust the parameters to taste.
5. Select Voice Processing from the Tree for the preset you are using. The Voice Page appears.
6. Find an unused Cord and Set it up like this:



7. Make sure the Wet/Dry Mix in the Amplifier section is set to 100 (all wet).
8. Play the keyboard soft and hard to check it out.
9. To reverse the action (hard = echo; soft = no echo), make the Cord amount +100 and set the Wet/Dry Mix to 0. (You may also want to reduce the Wet/Dry mix in the Echo effect itself to obtain the proper response.)
10. Now that you've got it all set up, try changing the Preset Effect. Try reverb, flanger, phaser and distortion effects.

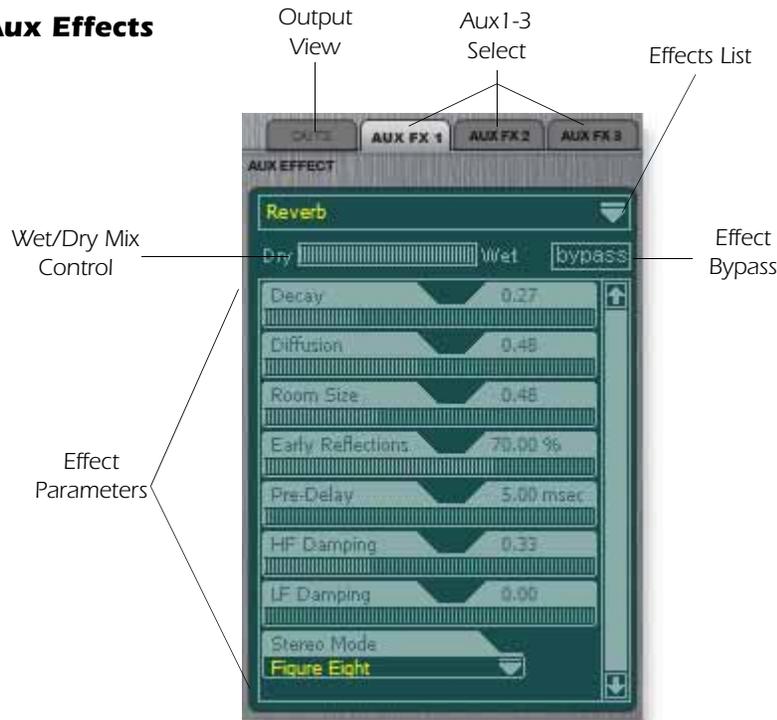
▼ Remember that if a preset with effects is used on more than one MIDI channel, the CPU penalty multiplies each time the preset is used.



Aux FX Screen

All effects have a wet/dry mix parameter to control the ratio of effected-to-plain signal, which is stored with the FX preset. The effect parameters vary with the type of effect.

Generally, when using the Aux Effects, the wet/dry mix in the effect should be set to 100% wet since the Aux Send amount controls how much effect will be applied.



Effects List	Click on the Effects List icon ▼ to show the list of effects.
Aux 1-3 Select	Select between Aux 1, Aux 2, or Aux3 FX in the TV view.
Output View	Selects the Main Output Window. See Main Output Window . (Note: This button is disabled in “Single” mode.)
Wet/Dry Mix Control	Adjusts the ratio of effected signal to un-effected signal.
Effect Bypass	When this button is selected (Red), the effect is completely bypassed, allowing you to hear only the unaffected signal.
Effect Parameters	Each effect has its own set of user adjustable controls.

Aux FX Send Amounts

The Aux Send Amounts can be controlled either from the Preset Cords or from within each Voice. Please refer to the FX Routing Diagram on [page 234](#). Aux Effect parameters, such as Delay Time, Feedback, etc., can not be externally controlled.

Preset FX Screen

All effects have a wet/dry mix parameter to control the ratio of effect-to-plain signal, which is stored with the FX preset. Each effect has its own set of control parameters. Turn up the Wet/Dry mix to hear more effect while you adjust the parameters, then turn the amount down to the desired effect amount.

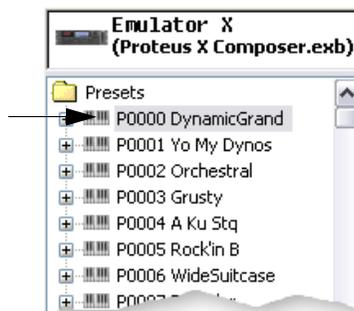
Preset Effects



Effects List	Click on the Effects List icon ▼ to show the list of effects.
FXA / FXB Select	Select between FXA or FXB in the TV view.
Wet/Dry Mix Control	Adjusts the ratio of effected signal to un-effected signal.
Post FX Send Amounts	Adjusts the amount of preset effect signal that will be sent to the Main and Aux Outputs.
Effect Parameters	Each effect has its own set of controls.

► To Get to the Preset FX Screen

1. Select the preset you want in the tree by clicking on the keyboard  icon. The Preset Global page appears.
2. Click on the Effect Select Icon ▼ at the top right corner of the Effects TV screen. A list of available effects drops down.



► To Swap FXA and FXB

Effects can sound very different depending on their order in the chain. Here is a simple way to swap FXA and FXB.

1. Choose **Swap Effects** from the Preset pull-down menu.
2. The two effects will exchange positions.

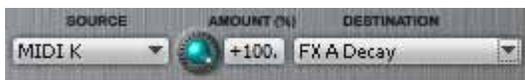
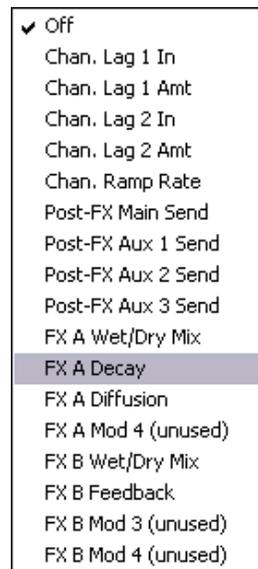
Preset FX Modulation Parameters

Certain parameters in each Preset Effect can be controlled from the Preset Cords. Some effects (such as BPM delay) allow four controllable parameters, while other effects (such as Early Reflections) have only one controllable parameter. If an effect uses less than four parameters, the idle cords are listed as **(unused)**. See below.

The Preset Cord value ADDS to the setting made in the Preset FX screen. For example, if the Wet/Dry Mix were set to 0% and you patched a MIDI Controller to Wet/Dry Mix with an amount of +100, the MIDI Controller would have a full range of control over Wet/Dry Mix. To invert the control, simply set the initial Wet/Dry Mix to 100% and set the Cord amount to -100.

► How to Connect MIDI Controllers to FX Parameters

1. Select the preset you want in the tree by clicking on the keyboard  icon. The Preset Global page appears.
2. Assign FXA or FXB in the Preset FX screen.
3. Select the **Destination** field from the Preset Cords section. A list of all possible Preset Cord destinations appears.
4. The effect modulation parameters are at the bottom of the list. Select the parameter you wish to control.
5. Select the **Modulation Source** you wish to use.



6. Set the **Cord Amount**.
7. In the screen above MIDI Controller K is controlling the Reverb Decay Time.

See [MIDI Channels & Real-time Controls](#) to learn how MIDI controllers work in Emulator X.

See [page 28](#) to learn how to assign MIDI controller numbers to Emulator X's A-P reference letters.

Creating and Deleting Templates

You can save a library of your favorite effects as *Templates*. Templates are saved in the Emulator X program files, so they're always available for use. For more information on templates, see ["Templates" on page 279](#).

Effects Listing

Reverb	Delay (mono)	Phaser
Early Reflections Reverb	Early Reflections	Pitch Shifter (mono)
Reverb Lite (mono)	EQ - 1 Band Parametric	Ring Modulator
Chorus	EQ - 4 Band	SP12-ulator
Chorus / Delay (mono)	Flanger	Tremulator BPM
Compressor	Flanger (BPM)	Tube
Delay	Growl	Twin (mono)
Delay (BPM)	Limiter	

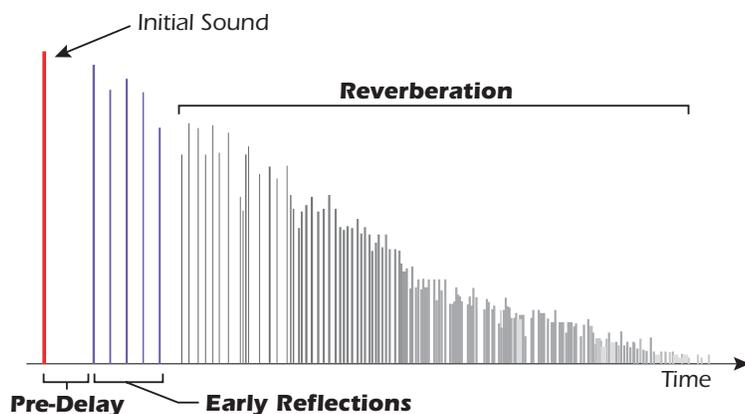
Effects Descriptions

Background: Reverb

This information applies to all the reverb algorithms.

Reverberation is a simulation of a natural space such as a room or hall. **Pre-delay** is the time before the reflections from the closest walls are heard. This parameter gives our ears important clues about the size of a room. As you might imagine, large halls have a longer pre-delay than small rooms.

Decay Time defines the time it takes for the reflected sound from the room to decay or die away. The diagram below shows a generalized reverberation envelope.



After a short pre-delay period, the echoes from the closest walls or ceiling are heard. The pattern of these first echoes, or **Early Reflections**, vary greatly depending on the type of room. Some time after the early reflection cluster ends, the actual **Reverberation** (a dense cloud of complex wall reflections) begins and decays according to the time set by the Decay parameter.

Diffusion is the amount of scattering and density of the reverberation cloud. Rooms with many complex surfaces have more diffusion than bare rooms.

High frequency energy tends to fade away first as a sound is dissipated in a room. The **High Frequency Damping** parameter adjusts the time it takes for the high frequency energy to die away and thus changes the characteristics of the room. Rooms with smooth, hard surfaces are more reflective and have less high frequency damping. Rooms filled with sound absorbing materials, such as curtains or people, have more high frequency damping.

Reverb

The Reverb is a medium load reverb, designed to simulate various types of rooms, halls and plates.

Parameter	Description
Decay	Sets the decay time of the entire effect.
Diffusion	Sets the amount of scattering of the reverb cloud. Note that high diffusion settings will increase the overall decay time.
Room Size	Changes the spacing of the reverb stages to simulate rooms of various shapes and sizes.
Early Reflections	Sets the volume of the initial wall reflections.
Pre-Delay	Sets the amount of time before any reflected sound is heard. Range: 1 millisecond to 200 milliseconds
High Freq. Damping	Sets the rate at which high frequencies die away.
Low Freq. Damping	Sets the rate at which low frequencies die away.
Stereo Mode	<p>Independent - In this mode, the left and right sides function as completely separate reverb units.</p> <p>Figure Eight - This mode incorporates criss-cross feedback in which the left channel feeds the right and vice-versa.</p> <p>Matrix - This mode utilizes multiple feedback paths to create a spacious, natural sound.</p>

◆ When using the Figure Eight or Matrix modes, pan position will be less focused and the stereo image will be somewhat blurred.

Modulation Parameters

- Wet/Dry Mix
- Decay
- Diffusion

Early Reflections Reverb

The Early Reflections algorithm is the most resource-intensive reverb model and has the most adjustable parameters. This reverb is designed to simulate various halls, rooms and reverberation plates. See also [“Background: Reverb” on page 243.](#)

Parameter	Description
Pre-Delay	Sets the amount of time before any reflected sound is heard.
Early Reflection Mode	Selects the pattern of the early reflections. Select between: Hall 1, Hall 2, Hall 3, Room, Spiral, 6 tap, 12 tap, 12 tap rising, or up & down.
Early Reflection Room Size	Scales the spacing of the early reflections to simulate rooms of different sizes.
Early Reflection L/R Spacing	Scales the spacing of left side reflections versus the right side, to create a more interesting stereo image.
Early Reflection / Tail Balance	Allows you to balance the relative volumes of the early reflections and the reverberation cloud. 0% to 100%
HF Damping	High Frequency Damping cuts the High Frequency content to simulate soft, absorbent surfaces such as draperies, wood or people.

Parameter	Description
LF Damping	Low Frequency Damping cuts the Low Frequency content to simulate hard, reflective surfaces such as tiles or stone.
Tail Decay	Sets the decay time of the reverberation cloud. (RT60)
Tail Diffusion	Sets the amount of scattering of the reverb cloud. At low settings, discrete echoes are heard. At high settings a diffuse wash of sound is heard. High diffusion settings will increase the overall decay time.
Tail Room Size	Changes the spread of the reverberation cloud echoes, simulating rooms of various sizes and shapes.
Tail Stereo Mode	<p>Independent - In this mode, the left and right sides function as completely separate reverb units.</p> <p>Figure Eight - This mode incorporates criss-cross feedback in which the left channel feeds the right and vice-versa.</p> <p>Matrix - This mode utilizes multiple feedback paths to create a spacious, natural sound.</p>

◆ When using the Figure Eight or Matrix modes, pan position will be less focused and the stereo image will be somewhat blurred.

Modulation Parameters

- Wet/Dry Mix
- Decay
- Diffusion

Reverb Lite (mono)

The Reverb Lite is a monophonic, slimmed down version of the Big Reverb. This effect only uses about 3/4 of the DSP resources of the Big Reverb, (and is smaller than even the Chorus effect) making it the perfect choice when you're running low on CPU horsepower. See also ["Background: Reverb" on page 243.](#)

Parameter	Description
Pre-Delay	Sets the amount of time before any reflected sound is heard.
Decay	Sets the overall decay time of the reverberation effect.
Diffusion	Sets the density of the echoes and scattering of the reverb cloud. At low settings discrete echoes will be heard, At high settings a dense wash of sound is produced. Note: High diffusion settings will increase the overall decay time.
Room Size	Changes the spacing between the reverb stages to simulate rooms of various shapes and sizes.
HF Damping	High Frequency Damping sets the rate at which low frequencies die away. A setting of zero results in no damping.
Pan	This parameter sets the position of the reverberant sound in the stereo field, in order to change the apparent source position.

Modulation Parameters

- Wet/Dry Mix
- Decay
- Pan

Chorus

Background: Chorus

An audio delay in the range of 15-20 milliseconds is too short to be an echo, but is perceived by the ear as a distinctly separate sound. If we now vary the delay time in this range, an effect called chorus is created, which gives the illusion of multiple sound sources. A slight amount of feedback serves to increase the effect. Inverted feedback percentages add negative feedback which can change the tone of the chorus. A very slow LFO rate is usually best for a realistic effect, but a faster LFO rate can also be useful with minimal LFO depth (.2).

This effect is a true-stereo chorus with independent delay lines for the left and right channels. The left and right modulation waveforms are always 180° out of phase, so that left channel delay is increasing while the right channel delay is decreasing and vice-versa. This creates a richer and more animated chorus sound.

Parameter	Description
Initial Delay	Sets the length of the delay. Range: 10ms to 50ms.
Modulation Depth	Controls the amount of LFO modulating the delay time. Increases the animation and amount of the chorus effect. Range: 0% to 100%
Modulation Rate	Sets the frequency of the low frequency oscillator. Range: .01Hz to 15Hz
Feedback	Sets the polarity and amount of delayed signal that will be recirculated through the delay lines. Range: -99% to +99%
LFO Waveform	Selectable between Sine or Triangle wave.

Modulation Parameters

- Wet/Dry Mix
- Rate (LFO)
- Feedback

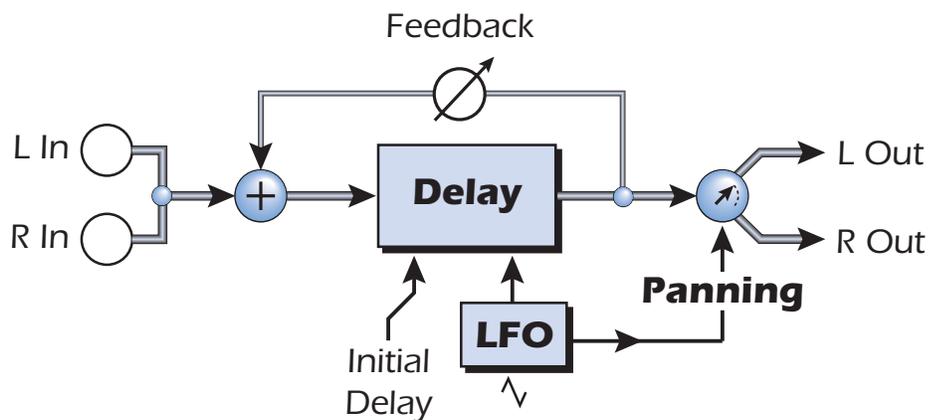
Chorus / Delay

This effect is a combination Chorus and Delay because the signal can be delayed up to a full second (1000 ms). Slightly modulating the delay time in the 15-20 millisecond range produces the familiar chorus effect. [See "Background: Chorus" on page 246.](#)

When a longer delay is modulated, a kind of gliding pitch shift effect is produced. Vibrato effects can be easily created by adjusting the modulation rate and modulation depth parameters. When feedback is introduced to a long modulated delay, the sound tends to get scrambled up in unpredictable ways which might be just the thing to create that "swarming bees" effect you've been looking for.

Setting the feedback percentage in the negative range, inverts the feedback signal. Negative feedback has the effect of deepening the cancellation notches which are created when using short delays with feedback.

- Very short delay times combined with a high feedback amount can be used to create monotone robotic-sounding effects.



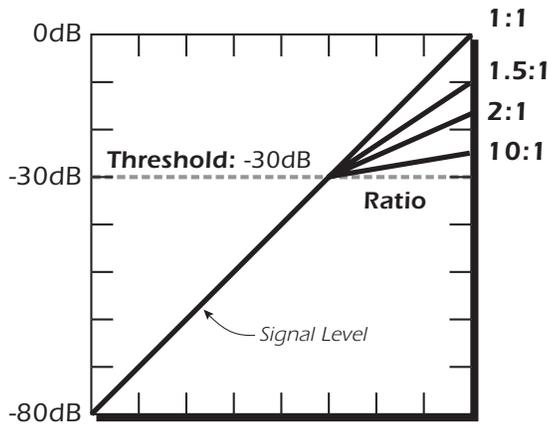
Parameter	Description
Initial Delay	Sets the length of the delay line. Range: 10ms to 1000ms.
Modulation Depth	Sets how much the LFO affects the delay time. Increases the amount of chorus effect and pitch shift animation. Range: 0% to 100%
Modulation Rate	Sets the frequency of the low frequency oscillator. Range: .01Hz to 15Hz
Feedback	Sets the polarity and amount of delayed signal that will be recirculated through the delay line. Range: -99% to +99%
Pan	This parameter moves the delayed sound left or right. This moves the apparent sound source left or right. Range: -99% to +99%
LFO -> Pan	Sets the amount of panning that will occur at the Modulation Rate. Range: -99% to +99%

Modulation Parameters

- Wet/Dry Mix
- Rate (LFO)
- Feedback
- Pan

Compressor

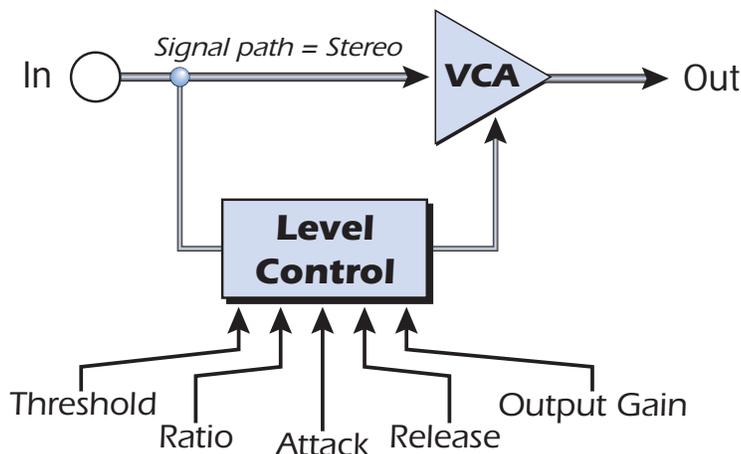
In its simplest form, an audio compressor is just an automatic gain control. When the volume gets too loud, the compressor automatically turns it down. Compressors are useful in musical applications because they smooth out dynamic volume changes and provide a louder average signal level.



Since the compressor turns down the gain of the signal, you might wonder how can it make the signal level stronger. An **Output Gain** control allows you to boost the output gain of the compressor in order to make up for the gain reduction. The overall level is higher and only turned down when the signal level gets too loud. The level at which the volume gets turned down is called the **Threshold**, which just happens to be the most important control on the compressor.

Basic Controls

The three main controls of a compressor are the **Ratio** control, the **Threshold** control and the **Output Gain** control.



If the signal falls below the *Threshold* level, no processing will take place. Signals exceeding the *Threshold* will have gain reduction applied as set by the ratio control. This important control allows you to dial in the range of amplitudes you want to tame. For example, if you're trying to trim off just the loudest peaks, set the threshold high so that compression only occurs during these peaks. **One of the biggest mistakes in using a compressor is having the threshold set too low.** This adds noise as the compressor will always be reducing the volume.

The *Ratio* control determines how strongly the compressor will affect the signal. The higher the ratio, the more reduction will be applied. If the ratio is high enough, (10:1) the signal will effectively be prevented from getting any louder. In this situation, the compressor will be acting as a *Limiter*, placing an upper limit on the signal level. In general, ratios from 2:1 to 6:1 are considered compression and higher ratios above 10:1 are considered limiting.

The *Output Gain* control amplifies the signal after it has been compressed to bring it back up in volume. If you don't increase the gain, the compressed signal will be much lower in volume.

Two other important controls are *Attack* and *Release*. Attack controls how quickly the gain is turned down after the signal exceeds the threshold. Release controls how fast the gain is returned to its normal setting after the signal has fallen below the threshold again. An attack setting of about 10 milliseconds will delay the onset of compression long enough to preserve the attack transients in guitar, bass or drums while allowing the sustain portion of the sound to be compressed. Longer release times are generally used to reduce the so called "pumping" effect as the compressor turns on and off. Don't make the release time too long, however, or the compressor won't have time to recover for the next pluck or hit. In general, the attack and release controls are used to smooth out the action of the compressor, but they can also be used to create special effects.

Parameter	Description
Threshold	Threshold sets the "trigger" level for compression. Signal levels above the threshold will be brought down in volume. Range: -50dB to +6dB
Ratio	Sets the ratio of input signal level to output signal level, or "how much" compression will be applied. Range: 0.20:1 to 10:1
Output Gain	Amplifies the signal after it has been compressed to bring up the volume. Range: -24dB to +12dB
Attack Time	Controls how quickly the gain is turned down after the signal exceeds the threshold. Range: 1 ms to 200 ms.
Release Time	Controls how fast the gain is returned to its normal setting after the signal has fallen below the threshold. Range: 0.2 second to 1.0 second

Modulation Parameters

- None

Delay

Background: Delay

A delay line makes a copy of the incoming audio, holds it in memory, then plays it back after a predetermined time.

Long delays produce echoes, short delays can be used for doubling or slapback effects. Very short delays can be used to produce resonant flanging and comb filter effects or create monotone robotic-sounding effects (Hint: use feedback). Stereo signals are summed together before entering a mono delay.

A feedback path recirculates the delayed audio back around through the delay line. When creating echo effects, the feedback controls how many echoes will be produced. With short delay settings, the feedback control acts as a *resonance* control, increasing the amount of comb filtering produced by the delay line. [See "Flanger" on page 256.](#)

A Tone control in the feedback path cuts some of the high frequency energy each time the audio goes through the delay line. This simulates the natural absorption of high frequencies in a room and can also be used to simulate tape-based echo units.

The Wet/Dry mix controls how loud the echoes are in relation to the original signal.

This Delay is a true stereo delay line with completely independent left and right channels right channels. This delay line features four different modes, which affect how the signals are routed and cross coupled.

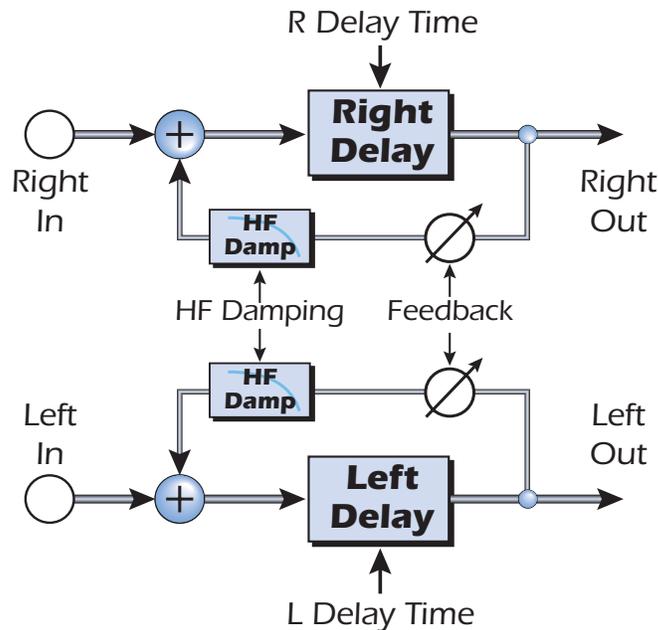
Discrete stereo

Use this mode when you want to preserve the stereo image of the delayed signals. You can also create panning effects by setting one delay long and the other short.

Delay Mode:

Discrete stereo

The left and right channels are kept totally separate.



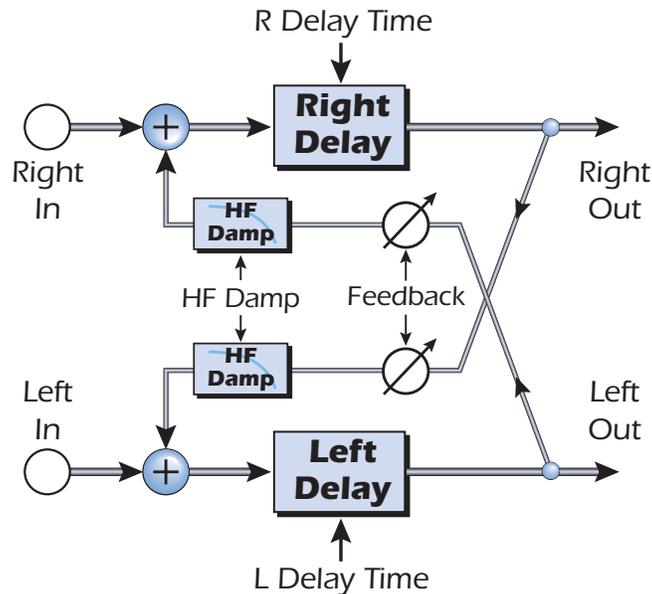
Ping Pong, stereo

This mode cross-couples the output of the left and right delay into the input of the opposite channel. This mode works best with true stereo program material or with completely different sounds on either channel.

Delay Mode:

Ping Pong, stereo

The output of the left and right channels are fed into the opposite channel.



Ping Pong, mono L->R

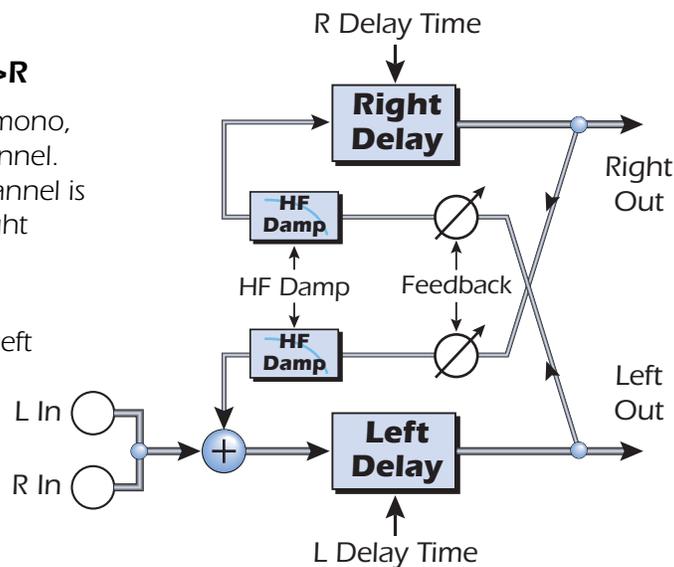
This mode creates echoes which bounce back and forth from left to right using either a mono or stereo source. This is the classic ping-pong echo effect.

Delay Mode:

Ping Pong, mono L->R

The input is summed to mono, then fed into the left channel. The output of the left channel is cross-coupled into the right channel.

The right channel output is cross-coupled into the left channel input.



Parameter	Description
Left Delay Time	Sets the length of the delay for the left channel in milliseconds. Range: .01 millisecond to 2000 milliseconds (.01ms. minimum increment between settings)
Right Delay Time	Sets the length of the delay for the right channel in milliseconds. Range: .01 millisecond to 2000 milliseconds (.01ms. minimum increment between settings)
Mode	Changes the signal routing of the delay. The four modes are: Discrete stereo - signal paths remain completely separate. Ping Pong, stereo - The feedback paths feed the opposite channel. Ping Pong, mono L->R - The effected signal is summed to mono, then fed down the left channel whose feedback path feeds the right channel. The right channel feedback path feeds the left channel. Ping Pong, mono R->L - This is the same as the previous algorithm except that the L-R channels are reversed.
Feedback	Sets the amount of delayed signal that will be recirculated through both delay lines. Range: 0% to 100%
HF Damping	High Frequency Damping cuts the high frequency content in the feedback paths. This causes the echoes to decay naturally away.

Modulation Parameters

- Wet/Dry Mix
- Feedback

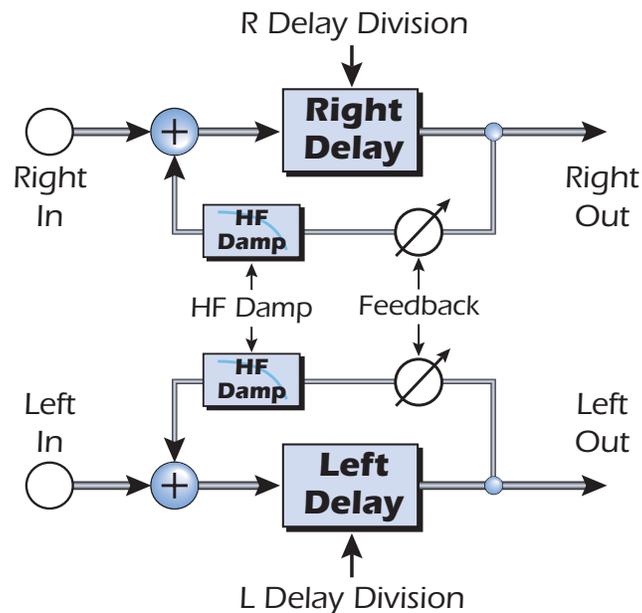
Delay (BPM)

The BPM Delay is similar to the Stereo Delay except that the delay time is synchronized to the master (MIDI) clock in note divisions. This allows you to easily set up synchronized echoes to your beats. Because the left and right channels are completely independent, you can set up different synchro-sonic patterns on each side. The four modes are the same as the stereo delay.

Delay Mode:

Discrete stereo

The left and right channels are kept totally separate.



◆ See [page 250](#) for full descriptions and diagrams. The BPM Tempo Chart on [page 135](#) shows the number of master tempo clock pulses for each division.

Parameter	Description
Left Delay Division	Sets the length of the delay for the left channel. Range: 4/1 note to 1/64 dotted-note
Right Delay Division	Sets the length of the delay for the right channel. Range: 4/1 note to 1/64 dotted-note
Mode	Changes the signal routing of the delay. The four modes are: Discrete stereo - signal paths remain completely separate. Ping Pong, stereo - The feedback paths feed the opposite channel. Ping Pong, mono L->R - The effected signal is summed to mono, then fed down the left channel whose feedback path feeds the right channel. The right channel feedback path feeds the left channel. Ping Pong, mono R->L - This is the same as the previous algorithm except that the L-R channels are reversed.
Feedback	Sets the amount of delayed signal that will be recirculated through both delay lines. Range: 0% to 100%
HF Damping	High Frequency Damping cuts the high frequency content in the feedback paths. This causes the echoes to decay naturally away

Modulation Parameters

- Wet/Dry Mix
- Left Delay Division (Changes the division by a power of 2, up or down.)
- Right Delay Division (Changes the division by a power of 2, up or down.)
- Feedback

Early Reflections

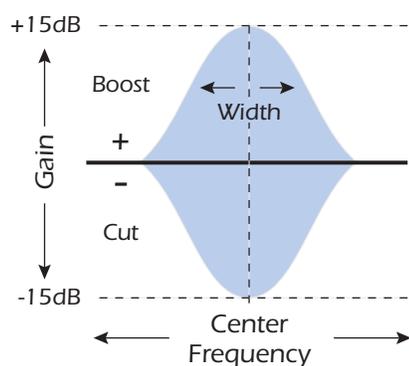
Early reflections are the initial echoes that you hear when listening to sound in an indoor space. The echoes are staggered because they are reflected off surfaces which are at various distances from your ear. Normally this effect would be part of a reverb algorithm, but we decided to include it as a separate effect because it creates useful and interesting sounds.

Parameter	Description
E.R. Mode	Selects the pattern of the early reflections. Select between: Hall 1, Hall 2, Hall 3, Room, Spiral, Multitap, 6 tap, 12 tap, 12 tap rising, up & down.
Room Size	Changes the spacing of the early reflections characteristics of various room sizes and shapes.
L/R Offset	Scales the spacing of left side reflections versus the right side, to create a more interesting stereo image. Range: -50% to +50%

Modulation Parameters

- Wet/Dry Mix

1-Band Para EQ



This single band parametric equalizer is useful when you just want to boost or cut a single range of frequencies. For example, if you want to brighten up a vocal, this EQ is ideal. This EQ offers up to +15dB boost and -24dB of cut.

Parameter	Description
Gain	Sets the amount of cut (-) or boost (+) of the selected frequency band. Range: -24dB to +15dB
Center Frequency	Sets the range of frequencies to be cut or boosted with the Gain control. Range: 40Hz to 16kHz
Bandwidth	Sets the width of the frequency range for the Center Frequency band that will be cut or boosted by the Gain control. Range: .01 octave to 2 octaves

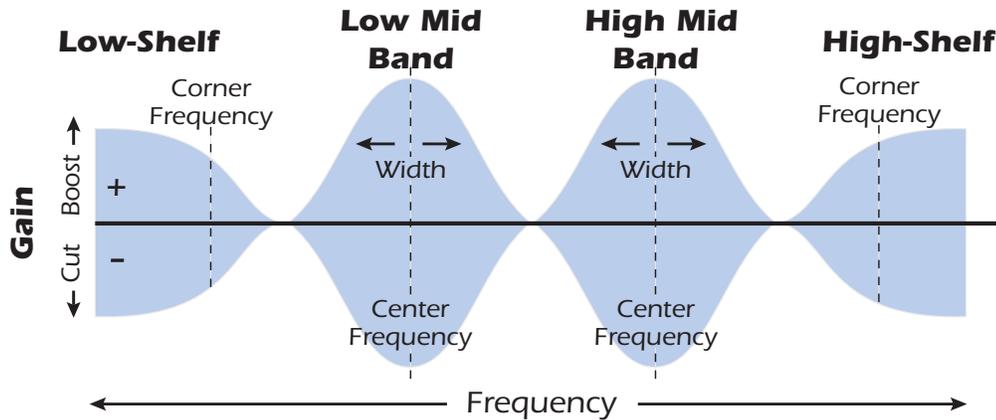
Modulation Parameters

- Gain
- Frequency
- Bandwidth

■ **Tip** - EQ becomes a powerful performance tool when the modulation parameters are patched to MIDI Controllers.

4-Band EQ

This 4-band equalizer provides two shelving filters at the high and low ends of the frequency range and two fully parametric bands in the center. Up to +15 dB of boost and -24dB of cut is provided for each band.



◆ **Note:** The Wet/Dry Mix control on an equalizer should normally be set to 100% wet or unpredictable results may occur.

Parameter	Description
High Gain	Sets the amount of cut (-) or boost (+) of the high frequency shelf. Range: -24dB to +15dB
High Frequency	Sets the corner frequency where the signal begins getting cut or boosted with the High Gain control. Range: 4kHz to 16kHz
High Mid Gain	Sets the amount of cut (-) or boost (+) of the High Mid Frequency band. Range: -24dB to +15dB
High Mid Frequency	Sets the range of frequencies to be cut or boosted with the High Mid Gain control. Range: 1kHz to 8kHz
High Mid Bandwidth	Sets the width of the frequency range for the High Mid Center Frequency band that will be cut or boosted by the High Mid Gain control. Range: .01 octave to 2 octaves
Low Mid Gain	Sets the amount of cut (-) or boost (+) of the Mid 1 Frequency band. Range: -24dB to +15dB
Low Mid Center Freq.	Sets the range of frequencies to be cut or boosted with the Mid 1 Gain control. Range: 200Hz to 3kHz
Low Mid Bandwidth	Sets the width of the frequency range for the Low Mid Center Frequency band that will be cut or boosted by the Low Mid Gain control. Range: .01 octave to 2 octaves
Low Gain	Sets the amount of cut (-) or boost (+) of the low frequency shelf. Range: -24dB to +15dB
Low Frequency	Sets the corner frequency where the signal begins getting cut or boosted with the Low Gain control. Range: 40Hz to 800Hz

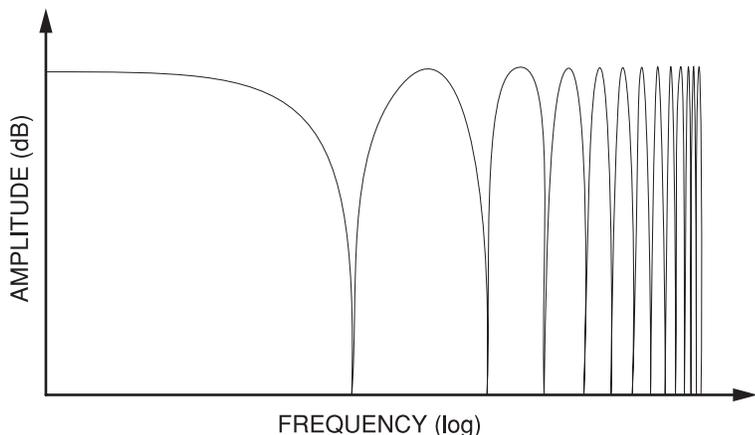
■ **Tip:** When using a MIDI controller to change the EQ gain, a value of 77 = 0dB.

Modulation Parameters

- High Gain
- High Mid Gain
- Low Mid Gain
- Low Gain

Flanger

A flanger is a very short delay line whose output is mixed back together with the original sound. Mixing the original and delayed signals results in multiple frequency cancellations known as a comb filter. Since the flanger is a type of filter, it works best with harmonically rich sounds.



A low frequency oscillator is included to slowly change the delay time. This creates a rich, sweeping effect as the notches move up and down across the frequency range. The amount of feedback deepens the notches, intensifying the effect. You can invert the feedback signal by choosing a negative feedback value. Inverting the feedback signal creates peaks in the notch filter and deepens the effect.

■ In order to hear a rich flanging effect, the original signal must be mixed in with the flanged signal. This can be accomplished using the Wet/Dry Mix control or with the Main Send (when using Aux effects).

Parameter	Description
Initial Delay	Sets the initial delay of the flanger in .01 millisecond increments. This parameter allows you to "tune" the flanger to a specific frequency range. Range: .01 ms to 4ms
Depth	Sets how much the LFO affects the delay time. Increases the animation and amount of the flanging effect. Range: 05 to 100%
Rate	Sets the speed of the low frequency oscillator which modulates the delay time of the flanger. Range: .01 Hz to 20Hz
Feedback	Controls how much signal is recirculated through the delay line and increases resonance. Negative values can produce intense flanging with some signals. Range: -99% to +99%
Through Zero	On - Adds a short delay to the original signal to simulate the classic "through-zero" flanger sound, which was originally created using two tape recorders. On (out of phase) - Inverts the phase (180°) and adds a delay to the original signal producing additional phase cancellation. If the feedback control is set to zero, complete cancellation occurs when the flanger goes through zero

Modulation Parameters

- Wet/Dry Mix
- Depth (Mod)
- Rate (Mod)
- Feedback

Flanger (BPM)

The BPM Flanger is similar to the previously described Flanger except that the delay time can be synchronized to the master (MIDI) clock in various note divisions. This feature makes it easy to synchronize sweeping flange effects to the beat of your song. In addition, the LFO sweep can be manually synchronized, using a continuous controller or switch in the Preset Cords, so that the top of the triangle waveform starts on a certain beat.

As the low frequency oscillator (LFO) varies the delay time, a rich, sweeping effect is created as the cancellation notches move up and down across the frequency range. The amount of feedback deepens the notches, intensifying the effect.

You can invert the feedback signal by choosing a negative feedback value. Inverting the feedback signal creates peaks in the notch filter and deepens the effect.

■ In order to hear a rich flanging effect, the original signal must be mixed in with the flanged signal. This can be accomplished using the Wet/Dry Mix control or with the Main Send (when using Aux effects).

Parameter	Description
Initial Delay	Sets the initial delay of the flanger in .01 millisecond increments. This parameter allows you to "tune" the flanger to a specific frequency range. Range: .01ms to 4ms
Depth	Sets how much the LFO affects the delay time. Increases the animation and amount of the flanging effect. Range: .006 to 100%
LFO Note Division	The speed of the LFO is locked to a note division of the master tempo. Range: 4/1 note to 1/64 dotted-note
Feedback	Controls how much signal is recirculated through the delay line and increases resonance. Negative values can produce intense flanging with some signals. Range: -99% to +99%
Through Zero	On - Adds a short delay to the original signal to simulate the classic "through-zero" flanger sound, which was originally created using two tape recorders. On (out of phase) - Inverts the phase (180°) and adds a delay to the original signal producing additional phase cancellation. If the feedback control is set to zero, complete cancellation occurs when the flanger goes through zero.
Sync (LFO Phase)	Re-syncs the LFO (to the peak of the triangle wave) whenever this button is pressed or whenever the sync modulation input goes from zero to a positive value.

Modulation Parameters

- Wet/Dry Mix
- Sync
- Rate (Note Div)
- Feedback

Growl

This is a new kind of self-modulation effect created especially for Emulator X. Growl excels at turning simple sounds into musically complex and interesting ones.

Depending on the input wave and control settings, Growl can impart a horn-like quality, a warm tube distortion, or a harsh rasp.

Growl is easy to use. Just turn up the depth control and wet/dry mix, then adjust the Initial control until you find the right amount of "growl". Then, back off the other controls and a fine tune all the other controls to taste. It's Grrreat!

Parameter	Description
Initial	This control "Tunes" the effect to a specific frequency range. Range: .01ms to 15ms
Depth	Sets how much self modulation is applied. Range: 0% to 100%
Color	Controls the tonal color of the effect. Range: -99% to +99%
Pre-Filter	Anti-aliasing filter. Turn this filter up if you hear aliasing at high-frequencies.

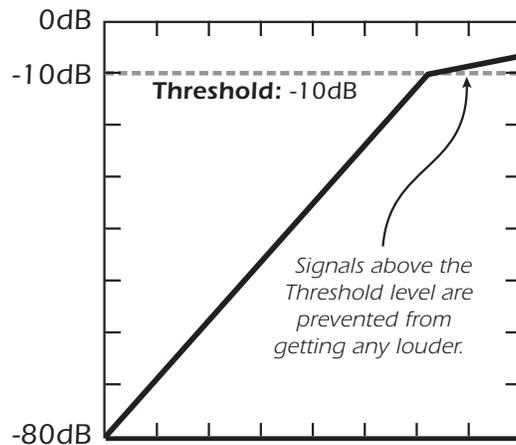
Modulation Parameters

- Wet/Dry Mix
- Depth
- Color

Limiters

The Limiter sets a “maximum level” on the audio. Beyond this *Threshold*, any increase in gain is effectively prevented or limited, hence the name. A limiter works like a compressor with a very high ratio setting.

The limiter is particularly useful for taming the highly resonant peaks that can be produced by the Z-plane filters. Set a fast attack time to suppress transients and use the Release control to reduce “pumping” effects.



The *Output Gain* control amplifies the signal after it has been limited to bring it back up in volume. If you don’t increase the gain, the signal may be lower in volume.

Two other controls are *Attack* and *Release*. Attack controls how quickly the gain is turned down after the signal exceeds the threshold. Release controls how fast the gain is returned to its normal setting after the signal has fallen below the threshold again. Longer release times are generally used to reduce the so called “pumping” effect as the compressor turns on and off. Don’t make the release time too long, however, or the compressor won’t have time to recover for the next pluck or hit. In general, the attack and release controls are used to smooth out the action of the limiter, but they can also be used to create special effects.

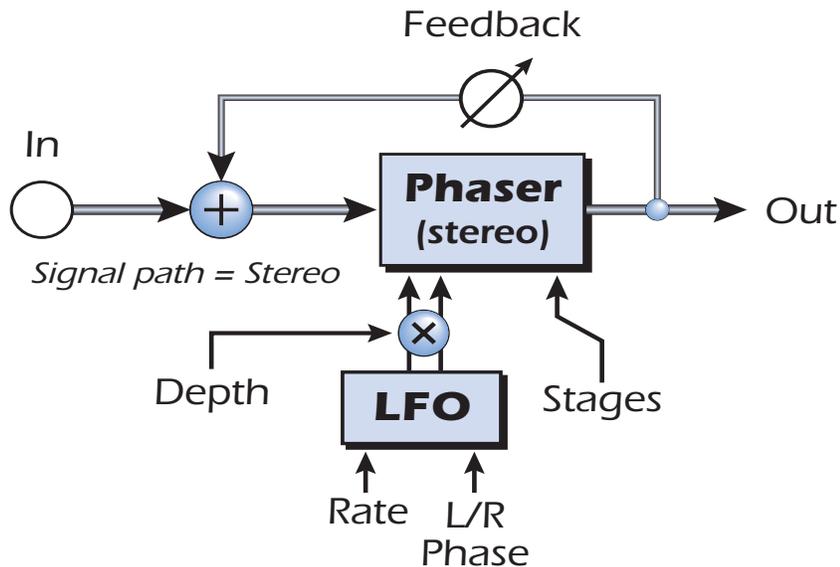
Parameter	Description
Threshold	Threshold sets the “trigger” level for limiting. Signal levels above the threshold will be brought down in volume. Range: -50dB to +20dB
Output Gain	Amplifies the signal after it has been compressed to bring up the volume. Range: -24dB to +12dB
Attack Time	Controls how quickly the gain is turned down after the signal exceeds the threshold. Range: 1 ms to 200 ms.
Release Time	Controls how fast the gain is returned to its normal setting after the signal has fallen below the threshold. Range: 0.2 second to 1 second

Modulation Parameters

- None

Phaser

A Phaser (or phase shifter) produces a fixed number of peaks and notches in the audio spectrum which can be swept up and down in frequency with a low frequency oscillator (LFO). This creates a swirly, ethereal sound with harmonically rich sound sources of a type of pitch shift with simpler sounds. The phase shifter was invented in the 1970's and the characteristic sound of this device evokes emotions of that musical era.



Parameter	Description
Depth	Controls how much the Center Frequency is swept by the LFO. Range: 0% to 100%
Rate	Controls the sweep rate of the Low Frequency Oscillator. Range: .01Hz to 20Hz
Feedback	Increases the depth of the notches and height of the peaks. Inverted values invert the feedback polarity. Range: -100% to +100%.
Stages	Selects between: 3, 6, 9, or 12 phase shift stages and thus the number of peaks and notches in the frequency response.
LFO L/R Phase	Offsets the LFO between the left and right phaser channels. (0°, 90°, or 180°). This adds extra animation to the sound.

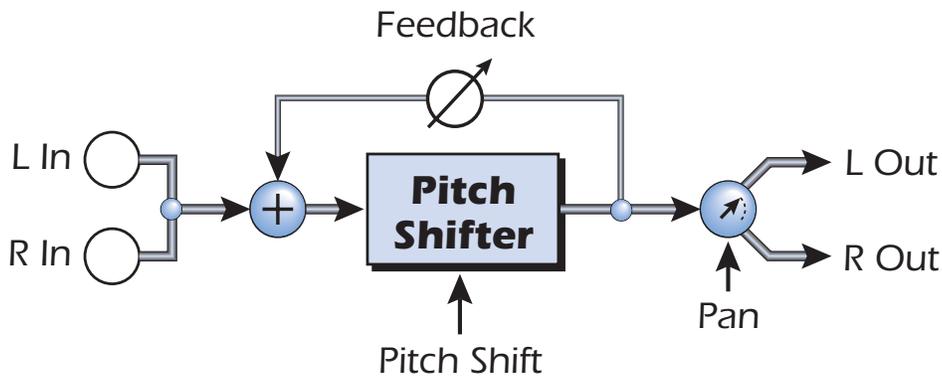
Modulation Parameters

- Wet/Dry Mix
- Depth (Modulation)
- Rate (Modulation)
- Feedback

Pitch Shifter (mono)

The pitch shifter (also known as a “harmonizer”) shifts the pitch of the input up or down while preserving the time between events. (Unlike speeding up or slowing down a tape recorder or pitch shifting across the Emulator X keyboard.) The range of the pitch shifter is -36 semi tones down to +24 semi tones up in .01 semitone increments.

Small amounts of pitch shifting makes an effective type of chorus. Semitone shifts work especially well on the human voice for harmonization and for special effects. With the Feedback control turned up, the pitch shifted (and slightly delayed) signal will be recirculated through the pitch shifter and re-shifted further up or down.



Parameter	Description
Pitch Shift	This control selects the pitch shift interval. Range: -36 semi tones to +24 semi tones
Feedback	Controls how much signal is recirculated through the pitch shifter. Range: 0% to 100%
Pan	This parameter moves the pitch shifted sound left or right. This moves the apparent sound source left or right.

Modulation Parameters

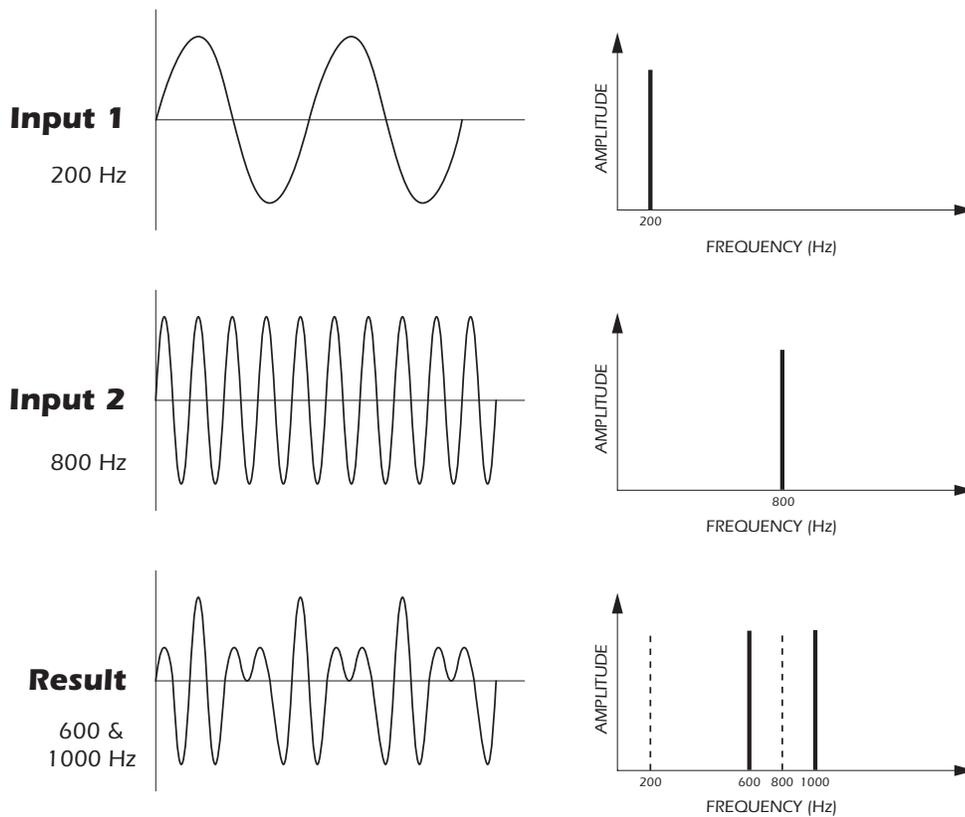
- Wet/Dry Mix
- Pitch
- Feedback
- Pan

Ring Modulator

A ring modulator takes two signals and multiplies them together to produce an output signal containing only the sum and difference frequencies of the two input waves. The original frequencies are not output. If complex waveforms are used, every partial of each signal multiplies every other according to its amplitude. Because ring modulation is a linear process, these new sum and difference frequencies are shifted from their normal exponential harmonic relationship. As a result, ring modulators tend to create clangorous, bell-like timbres.

In this ring modulator, each side of the stereo input source is multiplied by an internal sine wave oscillator. The internal oscillator can remain at a fixed pitch or it can be modulated by an envelope follower.

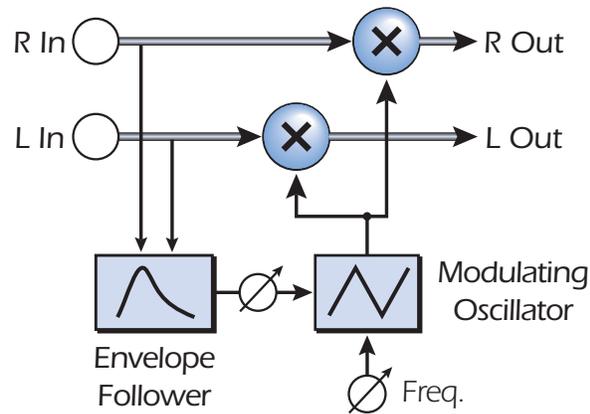
The envelope follower sweeps the pitch of the modulating oscillator up or down based on the amplitude of the input signal. Try adjusting the Emulator X attack and decay controls (amp) to hear the effect of the envelope follower. The envelope follower can sweep the modulating oscillator in either the positive or negative direction by setting the value positive or negative.



This diagram shows the result of Ring Modulating two sine waves with frequencies of 200 Hz & 800 Hz. Only the Sum and Difference frequencies of 600 Hz and 1000 Hz will be output.

Parameter	Description
Frequency	This control sets the frequency of the modulation waveform Range: 0 Hz to 5000 Hz
Envelope Follower	Controls how much the amplitude envelope sweeps the modulating oscillator. Range: -10x to +10x

Ring Modulator

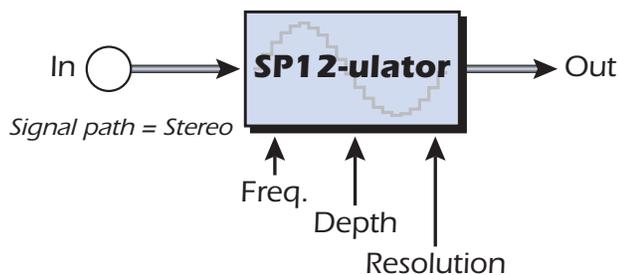


Modulation Parameters

- Wet/Dry Mix
- Frequency
- Envelope Amount

SP12-ulator

The E-MU SP-12 Sampling Drum Machine was the most advanced beat box of its day and still remains a classic. The SP-12 used a gritty, “drop-sample” pitch shifting technique which contributed to its distinctive sound. The SP12-ulator effect is an homage to the venerable SP-12, and allows you to mangle sounds as your forefathers did many years ago.



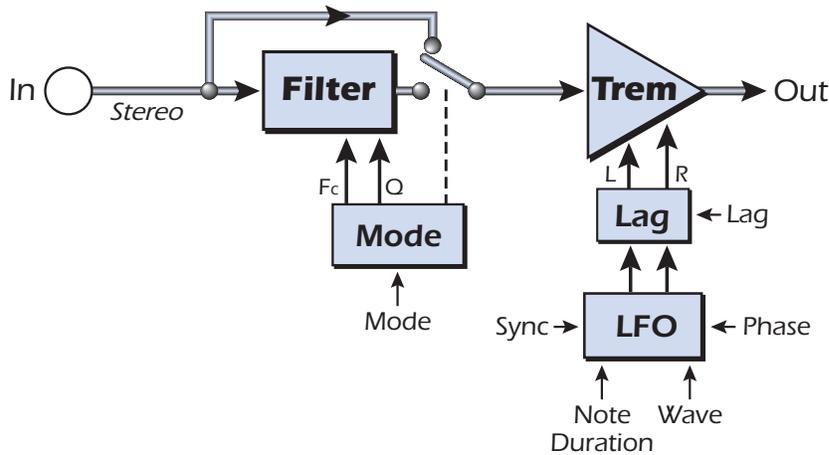
Parameter	Description
Frequency	This parameter controls the number of samples that will be “dropped.” Range: 0 - 100
Depth	This parameter controls the intensity of the resultant distortion. Range: 0% to 100%
Resolution	The SP-12 is a 12-bit drum machine. The SP12-ulator allows you to reduce the bit depth to 12-bits, or all the way down to 1-bit resolution if you must. Range: Hi-Res (32 bit), 12-bit to 1-bit

Modulation Parameters

- Wet/Dry Mix
- Frequency
- Depth

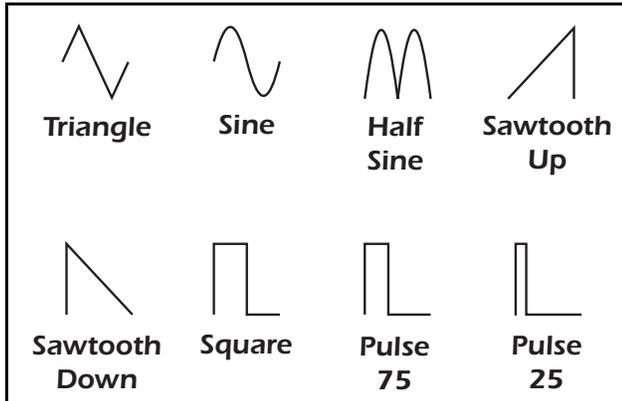
Tremulator BPM

Tremolo is amplitude modulation and is also a standard guitar amp effect. The Tremulator BPM is more like a tremolo on steroids! In addition to the standard rate and amount controls, the Tremulator lets you beat-sync the tremolo speed to the song tempo, has eight different modulation waveforms with adjustable lag, lets you autopan the stereo signal in several ways, and even includes a sweepable lowpass filter with five modes.



Parameter	Description
Wet/Dry	Controls the intensity (depth) of the tremolo effect. Range: 0% to 100%
LFO Note Duration	The speed of the LFO is locked to a note division of the master tempo. Range: 4/1 note to 1/64 dotted-note
LFO Waveshape	Selects between the following modulation waveforms: Sine, Triangle, Half Sine, Sawtooth Up, Sawtooth Down, Square, Pulse 25, Pulse 75
Lag	Sets the slew rate of the LFO output. This is normally used to round the corners of the square waves, but can create interesting effects with the other waveforms as well. Range: 0% to 100%
L/R Phase	Offsets the LFO phase between the left and right channels. (0°, 90°, 180°, 270°). 0° - Both the left and right channels are the same. 180° - Creates a back and forth, "ping-pong" effect. 90°/270° - Creates a rotating effect.
Mode	Six different modes are available: <ul style="list-style-type: none"> • Tremolo - Tremolo amplitude modulation • Tremolo + Filter 1 - Tremolo & 2-pole filter sweep, no Q • Tremolo + Filter 2 - Tremolo & 2-pole filter sweep, Mid Q • Filter 3 - 2-pole filter sweep, Low Q • Filter 4 - 2-pole filter sweep, Mid Q • Filter 5 - 2-pole filter sweep, High Q Note: The initial filter frequency is set lower as the Q is increased.
Sync (LFO Phase)	Re-syncs the LFO (to the peak of the wave) whenever this button is pressed or whenever the sync modulation input goes from zero to a positive value. You could connect a footswitch.

Tremulator Waveforms



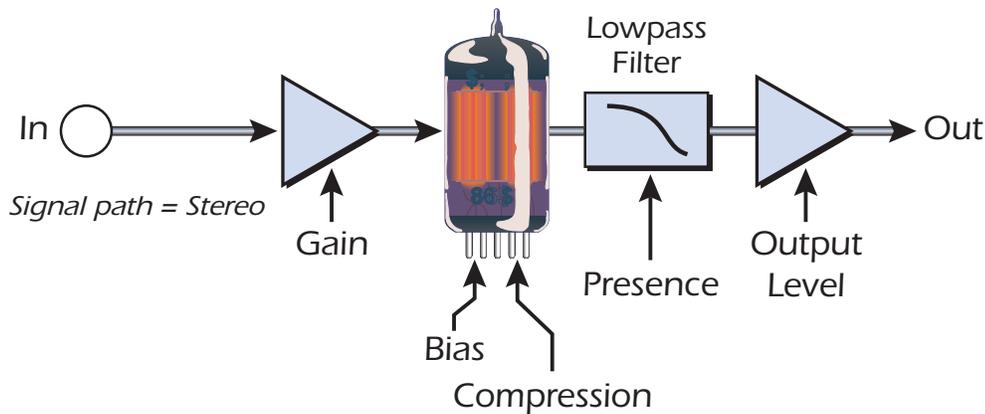
Modulation Parameters

- Wet/Dry Mix
- Rate (LFO Note Duration)
- Waveshape
- Sync

Tube

The Tube is a tube amplifier simulator based on “soft clipping”. An overdriven tube amplifier sounds good because it generates even harmonic distortion products. This digital tube gently rounds the waveform in a smooth, controlled manner much like a real vacuum tube.

Set the amount of distortion using the Gain control, then adjust the tone using the Treble and Bias controls. The Compression control smooths out volume changes and the Output Level control sets the final volume of the effect.



Parameter	Description
Gain	This parameter controls the overdrive amount on the “tube” and thus, the amount of distortion generated. Range: 0.25 - 50.00
Output Level	This parameter sets the output volume level and has no effect on the tone.
Presence	This control sets the lowpass filter frequency at the output of the “tube”.
Compression	Tubes have a certain amount of natural “soft” compression. This controls how much compression is applied.
Bias	Bias controls at what point in the waveform clipping begins.

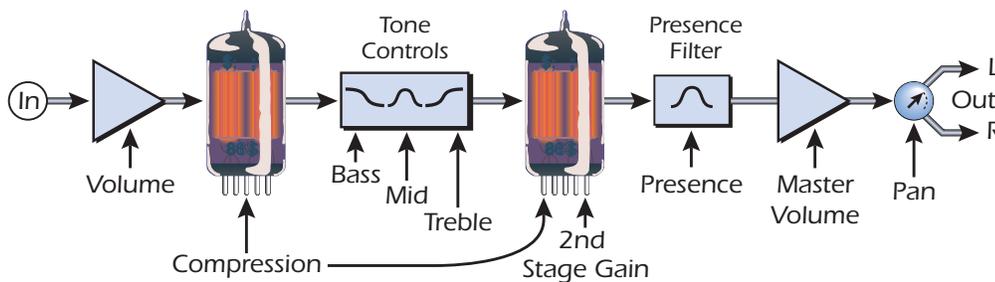
Modulation Parameters

- Wet/Dry Mix
- Gain
- Output Level

Twin

The Twin utilizes two “tube” stages to simulate the overdrive response of a guitar amplifier. Separate gain controls allow you to precisely control the amount of soft clipping from each “tube” stage. A three-band tone control between the two stages and a presence control after the 2nd stage allow you to precisely contour the overdrive frequencies.

Set the amount of distortion using the Volume and 2nd Stage Gain controls, then adjust the tone using the Treble, Mid and Bass controls. The Compression control smooths out the timbral changes and the Output Level control sets the final volume of the effect.



Parameter	Description
Volume	This parameter controls the overdrive amount on the 1st “tube” and thus the amount of distortion generated.
Master Volume	This parameter sets the output volume level and has no effect on the tone.
Treble	Allows you to adjust the high frequency content between the two “tubes”.
Mid	Allows you to adjust the mid frequency content between the two “tubes”.
Bass	Allows you to adjust the low frequency content between the two “tubes”.
2nd Stage Gain	This parameter controls the overdrive amount on the 2nd “tube” and thus, the amount of distortion generated.
Presence	Allows you to adjust the mid frequency content at the output of the 2nd “tube”.
Pan	Sets the left/right pan position of the processed sound. Range: -100% to +100%
Compression	Tubes have a certain amount of natural “soft” compression. This controls how much compression is applied by both tubes.

Modulation Parameters

- Wet/Dry Mix
- Gains 1 & 2
- Master Volume
- Pan

10 - Controls

This chapter describes how to work the controls Emulator X. There are certain tips and tricks to learn in becoming a power user, and a good number of them are in this chapter.

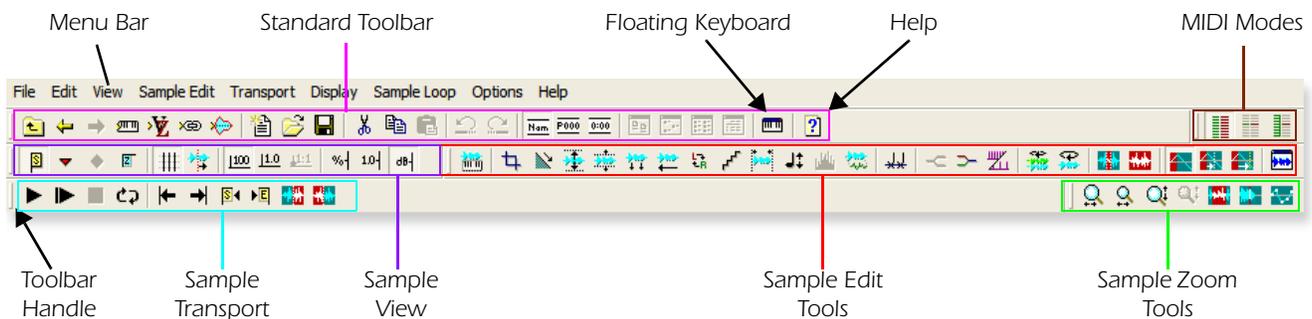
The Toolbars

The toolbars are located just beneath the menu bar at the top of the window. There is also a status bar at the bottom of the window. The toolbars provide handy single-click buttons for the most common operations. By holding the mouse pointer over the tool, a label will appear revealing the purpose of each tool.

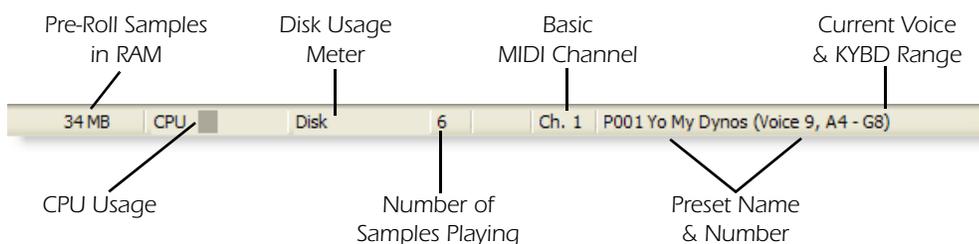
The upper toolbars can be rearranged by simply clicking the Toolbar Handles (see below) and dragging them wherever you want.

The Status Toolbar is always located at the very bottom of the window and provides context sensitive information, such as the current MIDI channel, sample/preset name & number, as well as the number of samples currently playing.

Menu Toolbars



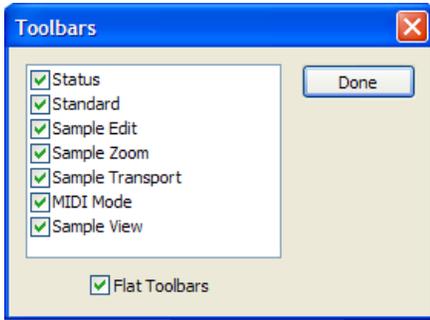
Status Bar



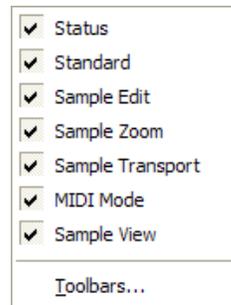
Viewing and Hiding the Toolbars

Select **Toolbars** from the **View** menu or Right-click when the cursor is over any toolbar to call up the Show/Hide Toolbar menu. Check the boxes of the toolbars you wish to view. "Flat Toolbars" removes the dividers and changes the look of the toolbars.

Select from the Menu Bar...



... or Right-click over the Toolbars



Drag & Drop

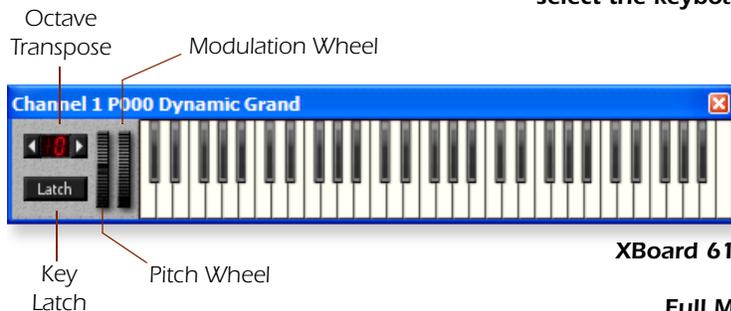
You can use simple "Drag & Drop" operations for many Emulator X functions. For example, to copy a voice to another preset, simply left-click on the voice icon, then drag it on top of another preset and release the mouse. To import WAVE, AIFF, REX2 or MP3 files, simply drop them onto the Emulator X icon. Page [328](#) provides a complete listing of the drag & drop functions.

Floating Keyboard

You can call up a movable keyboard window by selecting  **Keyboard** from the **View** menu, pressing **F9**, or by pressing the  button on the toolbar. The keyboard and wheels always send data on the current Basic channel with a default velocity of 64. When the basic channel is changed, all active notes will be cleared.

- **Transpose:** Shifts the keyboard up or down by up to 2 octaves. (No transpose on Piano and Full MIDI.)
- **Latch:** When latch mode is on, the note remains on when a key is pressed. Pressing the same key again turns the note off. Latched notes are displayed slightly darker.
- **Pitch & Mod. Wheels:** Send pitch bend and modulation wheel controls.
- **Keyboard Range:** Right-click anywhere on the keyboard to select the range. Choose the range that best matches your keyboard controller.

- Hold **Ctrl** and play the mini keyboard for a velocity of 10. Play the mini keyboard while holding **Shift** for a velocity of 127.



Right-click the keyboard to select the keyboard type.



Changing Settings

Entering Numeric Values

Numeric values are displayed in alphanumeric readouts with either an associated control knob or an up/down button adjoining the display.

► **Numeric values can be changed by:**

- **Turning the knob** (if there is one).
- Pressing the **Up/Down** keys when the control has the focus.
- **Typing** the desired value into the field.

► **Move the Cursor from the Computer Keyboard**

- Press the **Tab** key to move the cursor to the next field.
- Press **Shift+Tab** to move the cursor to the previous field.

► **Select the Knob or Slider and then:**

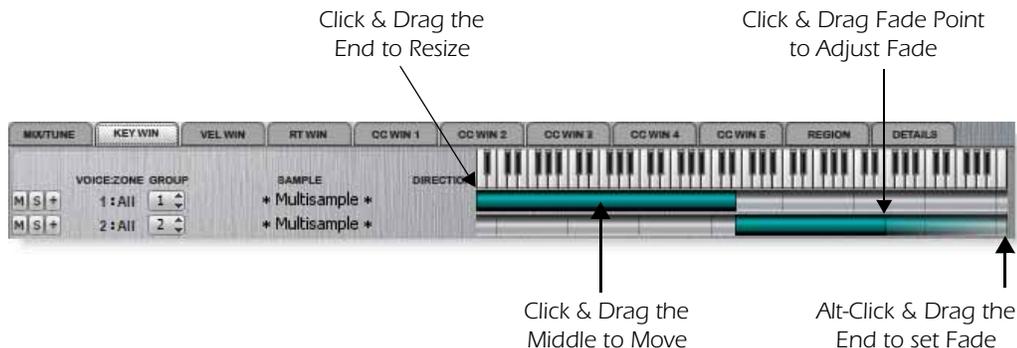
- Hold the **Alt** key and click the knob or slider to set the default value.
- Use the **Left/Right** keys to increment or decrement the value by a small amount.
- Use the **Page Up & Page Down** keys increment or decrement by a larger amount.
- Use the **Roller** on a roller-type mouse to increment or decrement the value.
- Press the **Home** key to turn the control all the way down.
- Press the **End** key to turn the control all the way up.
- Press the **5** key in the **numeric keypad** to center the control.
(Num Lock must be off)

Adjusting Ranges & Fades

The Key Win, Vel Win, RT Win, CC Win, and Region tabs in the Voices and Zones screen all contain parameters that can be adjusted by both graphical sliders and alphanumeric values.

You adjust a slider's range or position clicking and dragging your mouse. To create and edit a fade range, hold **Alt** key in conjunction with clicking and dragging the endpoint with your mouse. See below:

◆ **Note:** Fades are not available for the sliders in the **Region** tab.

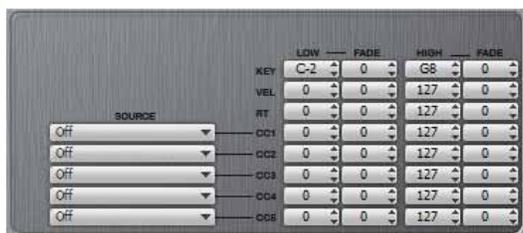


► Editing Ranges using the Graphic Display

- To set the high and low ranges, click and drag the end points of the graphic display
- To move the whole range without resizing it, click and drag the middle of the range.
- To create a fade, hold **Alt**, and click and drag the end points. Once you create a fade point, you can click and drag the fade point to re-position it.

► Editing the Ranges Using the Alphanumeric Fields

Alphanumeric Fields for Key, Velocity, Realtime, and CC ranges appear in the Overview section at the bottom of the Voices and Zones screen.



Select the voices you want to change, place the cursor inside the alphanumeric field, and then:

- Type in the desired value.
- Use the +/- keys to increment or decrement the value.
- Use the roller on a roller-type mouse to increment or decrement the value.
- Press and hold **Ctrl**, and press **Alt**. The word "MIDI" appears at the bottom of the window. Play your MIDI keyboard to set the key position.
(Note: Intelliedit must be turned on in the Preferences menu.)

Selecting Voices using your MIDI Keyboard

Before you begin: Make sure "IntelliEdit," in the MIDI Preferences, is set to the same MIDI port as your keyboard.

Press **Ctrl+Alt +MIDI Key** to select voices from the Voice Processing page. You'll see the word MIDI appear in the status bar at the bottom of the window.

The first note played selects all voices that overlap that key. The second note played, if it is different, selects all voices that overlap the range between the two notes played.

If the same key is pressed repeatedly, and if there are multiple voices assigned to that note, the selection will cycle through the voices whose ranges overlap that key.

You can see the selected range of keys displayed below the keyboard at the top of the Voice Processing window.

Adjusting Envelopes

Envelope rates and levels can be adjusted using the knobs, numeric entry, or by simply dragging the break points to the desired positions. See also, [“Amplifier, Filter & Auxiliary Envelopes” on page 132.](#)

BPM Mode & Tempo

BPM mode allows the envelope times to follow the internal/MIDI clock rate. The master tempo is adjusted and displayed in the Multisetup page ([page 74](#)). This allows the synthesizer to adapt to the tempo of your music.

The tempo syncs to an external MIDI clock if “External Tempo Source” is enabled in the Preferences menu. ([See “External Tempo Source” on page 27.](#)) If “External Tempo Source” is NOT enabled, the internal clock will be used.

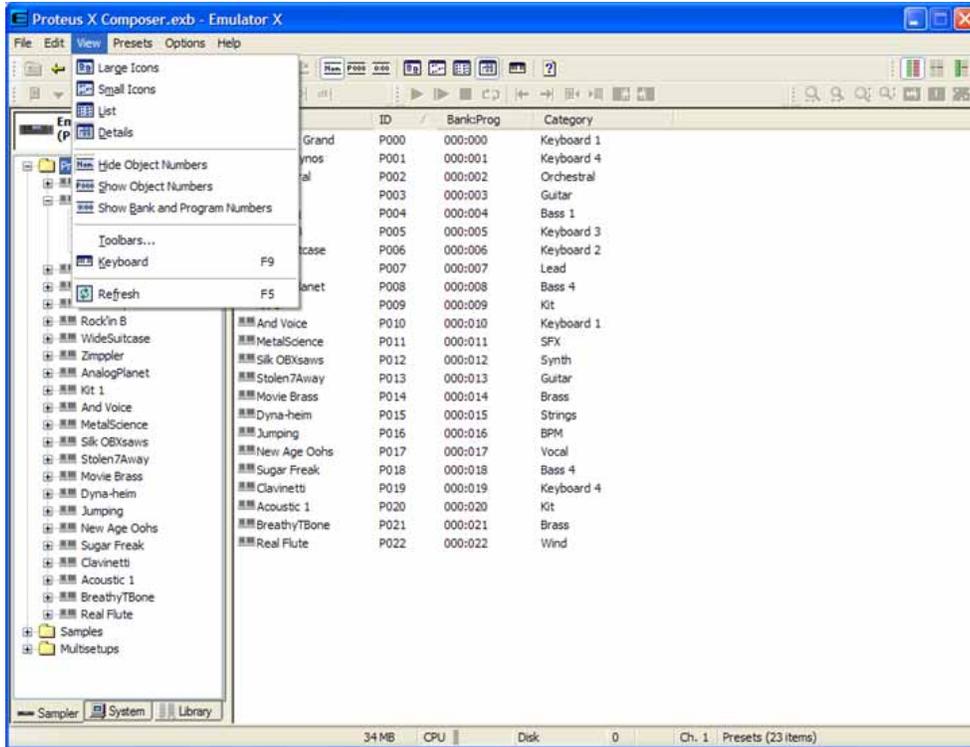
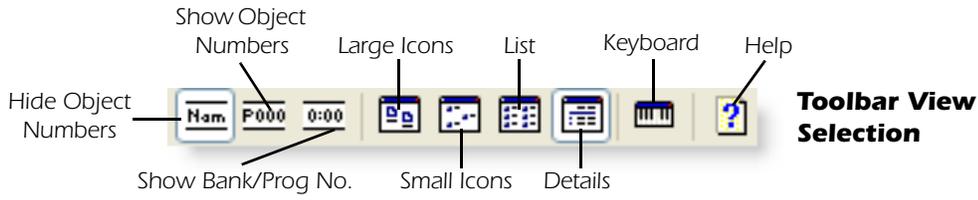


Cut, Copy & Paste

You can use cut, copy and paste commands for most Emulator X functions.

Page	Uses
Sample Page	<ul style="list-style-type: none"> ▪ Cut, Copy and Paste samples between banks. ▪ Cut or Copy, then Paste into a new sample (List View) ▪ Rearrange the order of Samples in the list.
Voices & Zones	<ul style="list-style-type: none"> ▪ Cut, Copy and Paste voices between banks. ▪ Rearrange the order of voices in the list.
Voice Processing	<ul style="list-style-type: none"> ▪ Cut, Copy and Paste synth module settings (such as envelopes, LFO, Cords) between voices in the bank. You can also copy Amp envelope settings to the Filter envelope for example.

View Menu



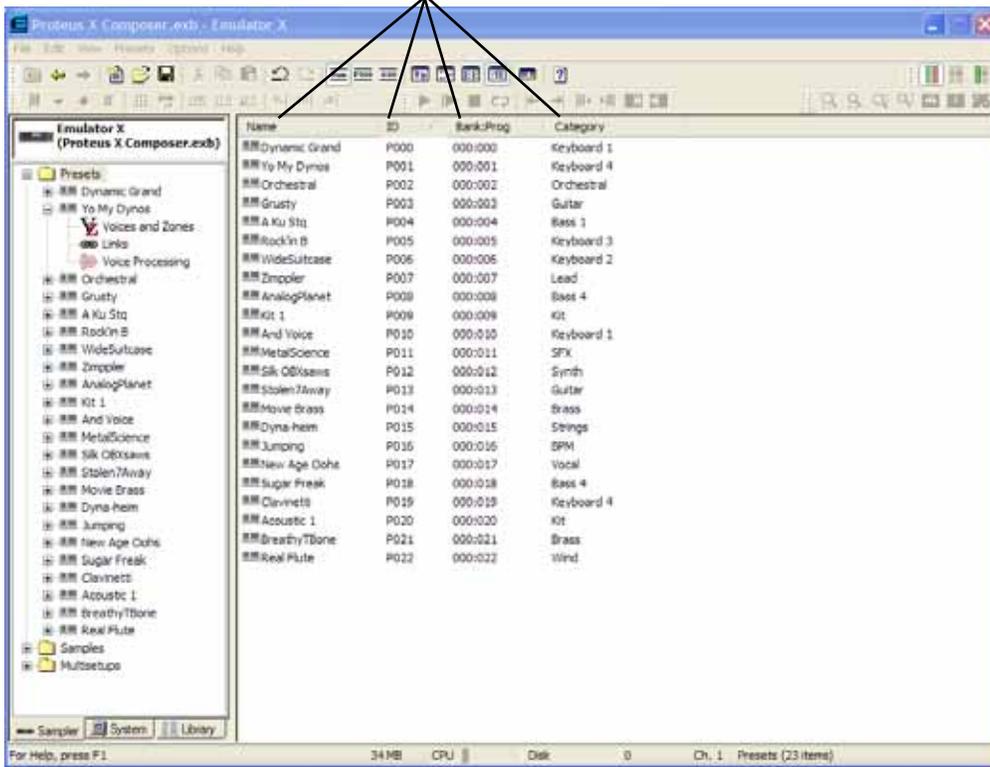
List Display View

When in the Tree, clicking on any folder shows the folder contents in the large window. The View Menu provides several display options for samples, presets and multisetups. The List Display view also allows you to cut, copy and paste multiple samples and presets. The view options are listed below:

- Large Icons** Displays large icons useful for smaller banks
- Small Icons** Displays small icons (shown above) useful for large banks
- List** Displays Presets and Samples in a list view format (in numerical order).

Details The Details view displays various details about the items. Clicking on the Column Header at the top of each column sorts the list by that category.

Category Buttons



Creating a Text List of Presets and Samples

You can create a text file of all the presets and samples in the bank. This can be useful to organize your sound collection. The preset list shows: preset number, bank/program number and preset name. The sample listing shows: sample number, sample rate, size in bytes, and the sample name.

To create a text file of the presets and samples in a bank, load the bank and select Save Text Listing... from the File menu.

Tree View

The Sample, Preset or Multisetup Tree items can also be viewed in several ways. The View Menu provides several display options for the tree. The view options are listed below:

Show/Hide Object Numbers

Displays or hides the object number for the Sample, Preset or Multisetup.

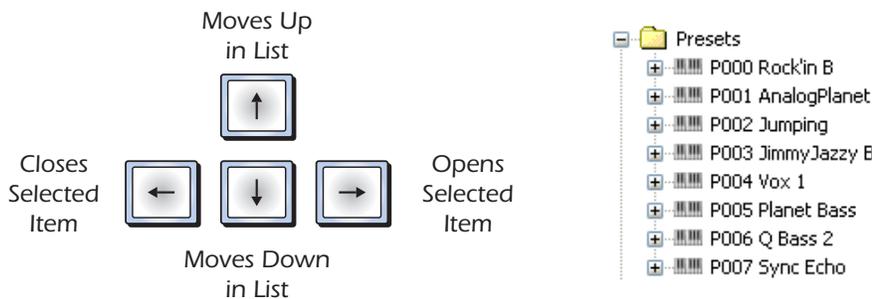
Show Bank & Program Numbers

Presets can be displayed in the tree with Object numbers (P003) or with Bank and Program numbers (002:003).

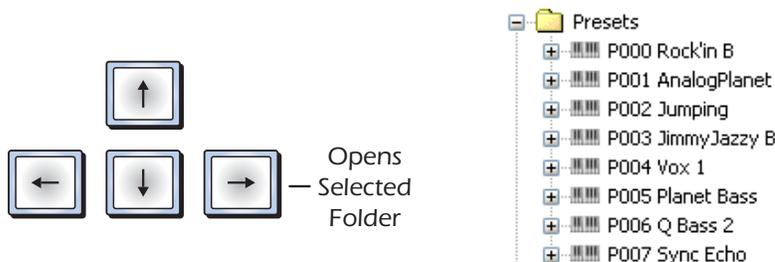
Tree Navigation using the Keyboard

Cursor Keys

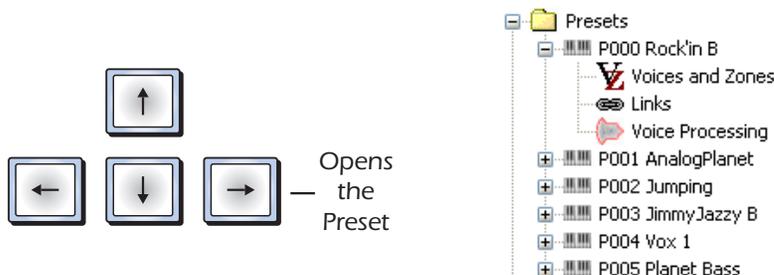
You can move quickly through the Tree using the Left-Right-Up-Down Cursor Keys.



With the Presets folder selected as in the example above, pressing the right cursor will open the presets folder.



Pressing the Right Cursor key again opens the Preset.



The Left Cursor reverses the actions. The Up-Down Cursors move up and down within the list. The Enter key opens the item.

Find Items

When the cursor is positioned in a list such as the tree, you can locate specific presets or samples by simply typing the name. Typing the first letter of a name finds the first preset or sample beginning with that letter. Repeatedly typing the same letter finds the next item beginning with that letter, and so on.

When “Show Object Numbers” is selected, you can type the preset or sample number (preceded by the letter P or S) to locate items by number.

Tree Navigation Using the Toolbar

You can use the buttons in the standard toolbar to navigate through the Tree.



Navigates up one Tree level.



Jumps back to the previous tree location.



Jumps forward to the next tree location if you’ve clicked the back button.



Jumps to the Global Presets page for the current preset.



Jumps to the Voices and Zones page for the current preset.



Jumps to the Links page for the current preset.



Jumps to the Voice Processing page for the current preset.

Refresh (F5)

Sometimes the display does not reflect the state of the synthesizer or vice-versa. Pressing F5 or selecting Refresh from the View menu manually synchronizes the display and synth.

Context Sensitive Menus

The right mouse button brings up context-sensitive menus in many areas of Emulator X. Simply right-click over the field or area to bring up the menu for that function.

All Sound Off (Ctrl + Pause)

Activating this function immediately terminates all sound from the synthesizer and clears the effect processors of all sound data. The All Sound Off command is automatically initiated when the sequencer transport **Stop** is pressed.

Templates

For each module in the Voice Processing window, the Preset window, or for Effects, you can save a library of your favorite settings. These Templates make it easy to create your own customized voices and presets. Templates are saved with Emulator X application so they're always available for use. Here's how they work:

Creating and Recalling Templates

You can create a template for a module, effects, all preset parameters, or all voice parameters to be recalled later. Templates are stored with Emulator X's program files and therefore can be recalled from any bank.

► To Create a Template from a Module or Effects

1. Set the controls as desired for a particular module or effect.
2. **Right-click** with the mouse anywhere inside the module border or Effects TV screen. A pop-up menu appears.
3. Select **Save Template...** or press **"S"** while the popup dialog box is visible. The following popup dialog box appears.
4. **Type in a name** for your Template, and click **OK**. The Template is saved and ready to use when you open another bank.



► To Save All Voice Parameters as a Template

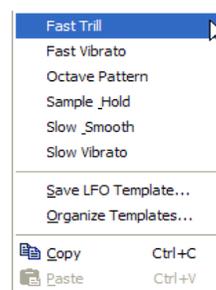
1. Go to the Voice Processing window.
2. From the **Voice Processing** menu, select **Voice Processing Templates**.
3. Choose **Save Voice Processing Template...** from the menu. A pop-up dialog appears asking you to name your new Template.
4. Name the template.
5. Press **OK** to save the Template or **Cancel** to cancel the operation.

► To Recall a Template

1. **Right-click** with the mouse anywhere inside the module border or Effects TV Screen. A pop-up menu appears.
2. Select the desired Template from the list. The Template will be selected.

OR...

1. Select **Organize Templates** or press **"O"** while the popup list is visible. The Organize Templates dialog box appears.
2. **Select the Template** you want, and click **OK**.



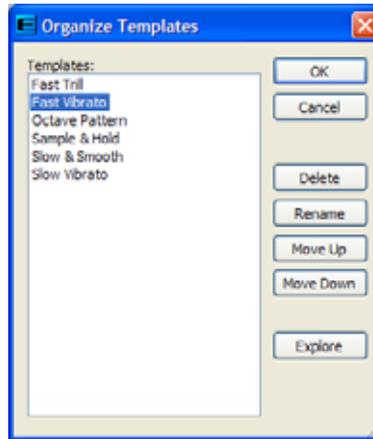
◆ The menu only shows the first 20 templates; if there are more than 20, an item labeled **More Templates** appears, which will bring up a dialog for selecting a template.

Organizing Templates

You can re-name, delete, re-order, and Explore your templates.

► To Rename or Delete a Template

1. Select the desired template from the Organize Templates dialog box.
2. Click **Delete** to permanently remove the Template.
3. Click **Rename**, then type in the new name to rename the Template.
4. Click **OK** to close the dialog box.



■ **Tip:** Several templates can be deleted in one operation by selecting multiple templates and clicking **Delete**. This also applies to moving **Templates Up** or **Down**.

► To Re-Order Templates

This feature allows you to select which templates appear at the top of the list, which is important if there are more than 20 templates.

1. Click the template you want to move.
2. Click either **Move Up** or **Move Down** to change the order of your templates.

► To Explore Templates Folders

1. From the Organize Templates dialog box, click **Explore**.
2. Explore the templates on your computer.

Emulator X stores your templates as .xml files in the following folder:

\Program Files\Creative Professional\Emulator X Family\Templates\

When you click **Explore**, you are taken to this directory. From here, you can copy or restore existing templates for backup purposes. You can also modify the .xml files with an XML editor.

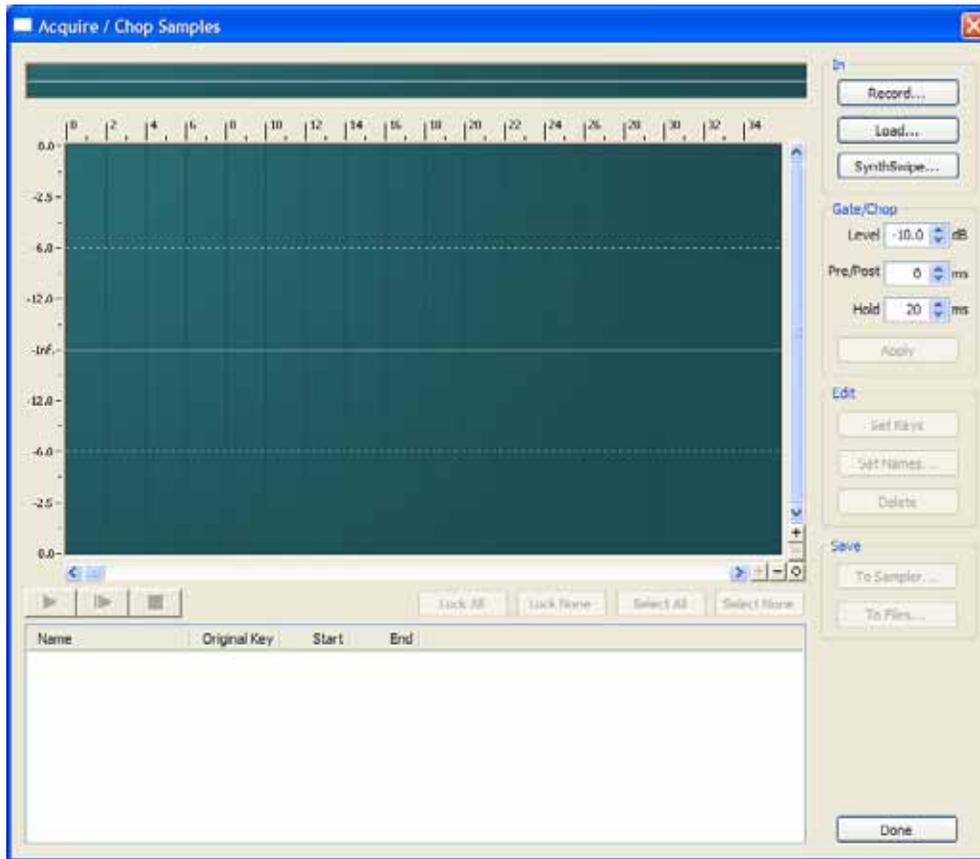
11 - From Sample to Preset

Acquire (Sampling)

Sampling has always been a bit of a chore to do because of all the housekeeping and organization involved. In Emulator X, we've invented a host of automatic tools to make sampling easier than ever before.

The chop function automatically extracts individual samples from a section of audio containing multiple notes. At the sample recording session, you set up your audio recorder and have the musician play each note separated by a short period of silence. You can make verbal cues on the tape if you wish. Later, when you're ready to input your samples, the periods of silence allows Emulator X to automatically separate the samples for processing.

Select **Acquire Samples...** from the File menu. The following screen appears.



Input Controls

Record	Select this option if you want to sample audio from an ASIO or DirectSound input.
Load	This option allows you to load Wave, AIFF, MP3, or REX 2 sample files to run them through the gating and sample placement options. Sample gating chooses samples which are separated by periods of silence.

■ To import other sample formats, see ["Translator Emulator X Converter" on page 319.](#)

Recording Audio (Sampling)

Choose "Record" if you want to sample from an ASIO source within your computer.

► To Set-up PatchMix DSP for Sampling into the Emulator X

Use these instructions if you are using PatchMix DSP. To sample from a PatchMix DSP input, an ASIO Send must be inserted on the desired PatchMix strip.

1. Open the PatchMix DSP mixer.
2. Add a mixer strip corresponding to the analog or digital input you are using. (i.e. If you have a microphone connected, add a mic/line strip.)
3. Insert an ASIO Insert Send into the mixer strip. You may also want to add a Peak Meter to make sure you're getting a proper input level.

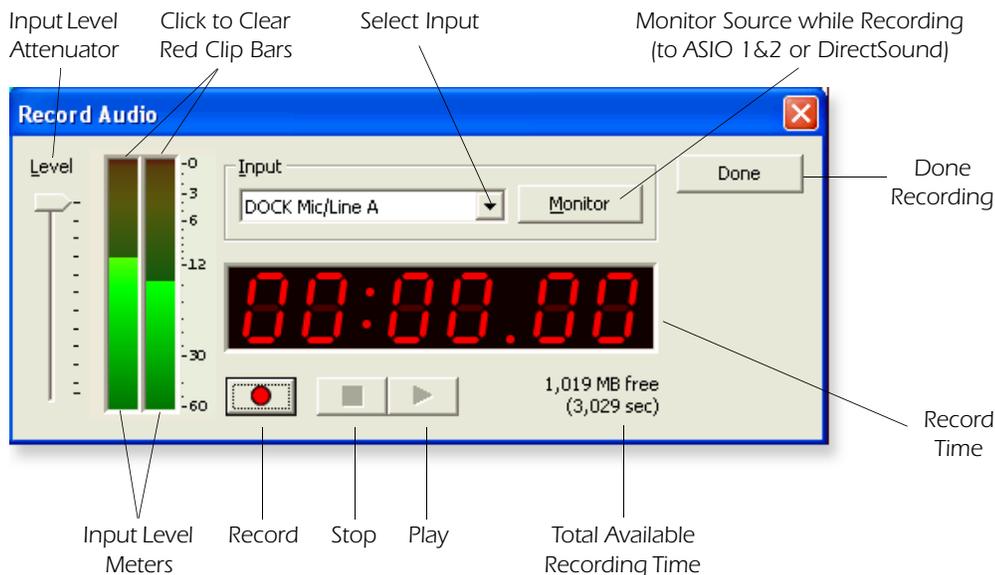
► To Set-up Emulator X for Sampling

4. Open the Acquire Sample dialog from the file menu on Emulator X.
5. Select Record from the input section in the upper right corner. The Record Audio dialog box (shown below) appears.
6. Click on the Select Input box. The open ASIO inputs will be shown. Select the input you wish to use.
7. You should see activity on the Emulator X record meters with audio on your input.
8. That's it!

Note: The Emulator X only records in stereo. To record monophonic samples, use a single input and convert the stereo sample to mono using the Mono<->Stereo sample edit tool. [See "Convert to Mono or Stereo" on page 194.](#)

► To Begin Sampling

1. Choose Acquire from the File menu. The Acquire/Chop Samples screen appears.
2. Press the Record... button. The following dialog box appears:



Record Audio Controls

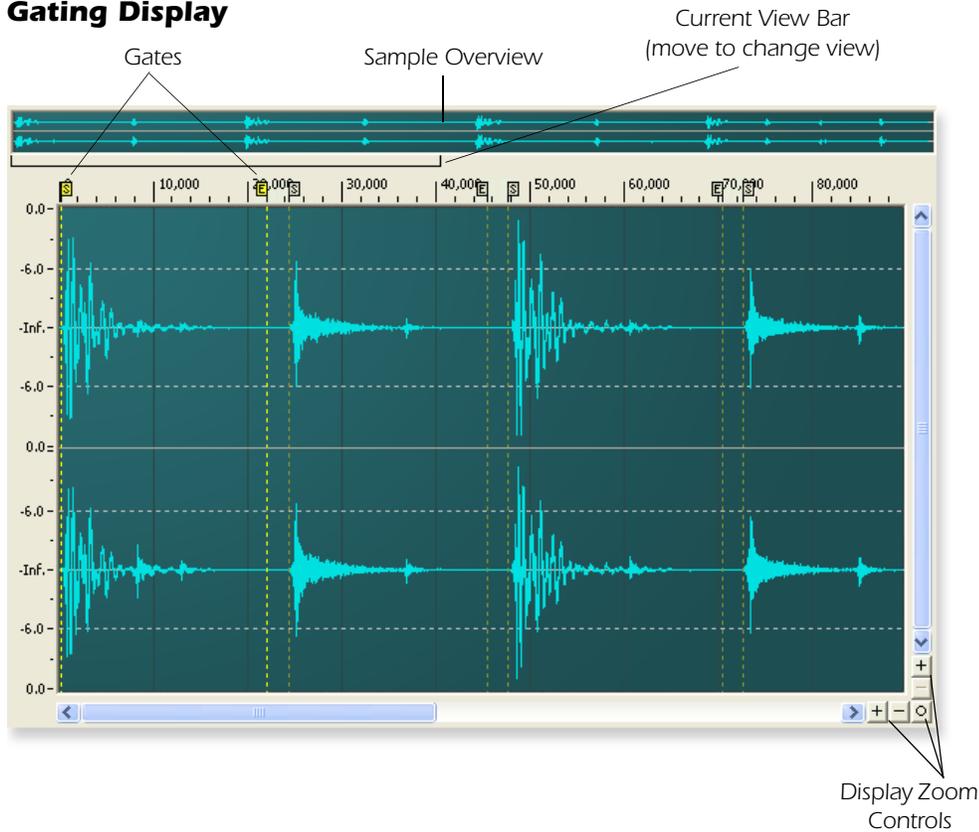
Level	This control allows you to attenuate the input signal to avoid clipping.
Input	Choose the desired input source. An ASIO source must be open before an input will be listed here.
Monitor	When this button is depressed, you can monitor the input before, during and after the sampling process. The monitor outputs through ASIO 1 & 2 (or DirectSound if selected). Monitor will normally be left On.
Record	Press to initiate recording. Recording continues until you press stop
Stop	Stops recording or playback of the sample.
Play	Plays back the recorded sample from the beginning. (Pressing the Space Bar toggles between Play and Stop.)
Done	Closes the Record window and returns you to the Acquire/Chop dialog with the new recording in.
Meters	Stereo input signal level. The top red bars stay on after clipping. Click to clear them.

3. Select the **Input** that you wish to record. The displayed input selections are based on which ASIO channels were free when the Record Audio dialog was opened.
 - **No Inputs?** If no inputs are listed, make sure E-MU ASIO is selected in the Preferences dialog. If you're using PatchMix DSP, make sure that the current Session contains at least one ASIO Send.
4. **Set your input levels.** With audio coming in, watch your meters. Adjust the input level using the Input Level Attenuator or by adjusting the volume level of the source. The signal peaks should illuminate the yellow area of the meter. The red bar at the very top of the meter indicates clipping and will stay on after clipping is detected. Left-click on either red bar to clear the clip indicator.
5. **Press the Record** button to begin recording. The Record Time display begins running and the Total Available Recording Time will begin to slowly decrease.
6. **Press the Stop** button (or hit the spacebar) when you're finished recording.
7. **Press Done** to return to the Acquire/Chop dialog. (Pressing Record again without clicking OK discards the previous sample.)
8. After pressing **Done**, you're ready to set up the automatic sampling features.

Gating the Samples

The sample gating feature is designed for separating a series of samples separated by silence (SSSS). You can record or import one long sample containing many short samples and let the gating feature automatically separate them for you.

Gating Display



■ Display Units

Right-click on the horizontal or vertical rulers to change the display units.

Horizontal units can be displayed in samples or seconds.

Vertical units can be displayed in percent, decimal, or dB.

Gate/Chop Controls

Level	Sets the level change that must occur before a gate is set. Adjust this control so that the gates conform to the signal. The gates should appear as in the diagram above.
Pre/Post	Sets the pre-sample and post-sample time. Increase this setting if the signal attacks and releases are being truncated.
Hold	Sets the minimum amount of time that must pass before another gate can be set. This control is useful to prevent too many gates from being set.
Apply	Sets the gate start and end points according the current Gate/Chop settings

► To Gate & Chop Samples

After recording or loading a section of audio, you're ready to gate and chop.

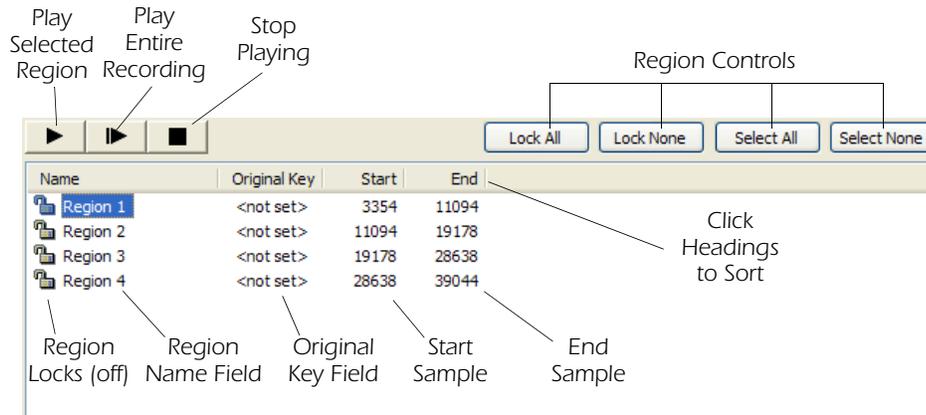


Setup Parameters

1. Set the threshold **Level** using the numeric display or by setting the threshold markers in the waveform display. This setting affects the accuracy of the gating. For percussive samples like the ones shown above, set this setting at -40 dB or lower to start. Different signals require different settings.
2. Set the Level, Pre/Post and Hold settings to the settings shown at left as a starting point.
3. Press the **Apply** button. Gates now appear on the waveform display. You'll also notice that numbered Region Names now appear in the Region List below the waveform display. We'll get to these in a minute.

■ You can also adjust the Threshold Level settings by simply dragging the markers. Hint: Zoom in first.

Region List Display



■ **Tip:** Select a Region and hit the Space Bar to play it.

4. The gates should conform to the individual samples, similar to the screen shown on [page 284](#). If not, simply change the Gate/Chop controls and **Apply** again. You will be asked if you want to "Delete the Existing Regions." Click **Yes**.

- **Decrease the Level setting** (bigger negative number) if the regions are too large.
- **Increase the Level setting** (smaller negative number) if the regions are too small.
- **Increase the Hold time** if too many gates are being set.

Setting the gates may take a little practice to master, but it's really very simple. In any case you can always set the gates manually if you wish. Read on.

Region List Controls

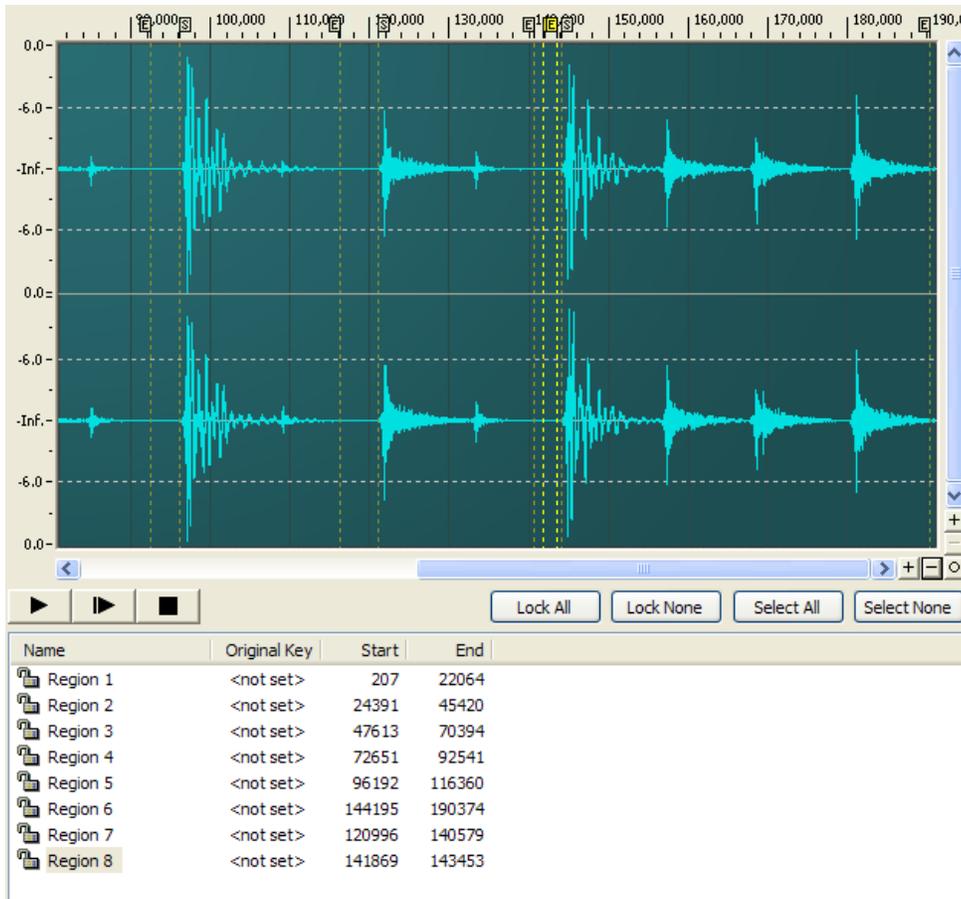
	Play - Plays the currently selected Region.
	Play All - Plays the entire Sample (all the Regions)
	Stop - Stops playing
Lock All	Locks all Regions.
Lock None	Unlocks all Regions
Select All	Selects All Regions
Select None	De-selects all Regions
Name Field	Displays Region name, Type directly into this field or use "Set Names" button.
Original Key	Displays key number of region. Type directly into this field or use "Set Keys" button to generate key numbers using pitch detection.
Start Field	Displays the Start point of the Region in samples.
End Field	Displays the End point of the Region in samples.
Lock Icon	Locked Regions are protected against modification. Click to lock or unlock.
Set Keys	Generates key numbers based on pitch detection of the region.
Set Names...	Allows you to enter a name for any selected Regions and automatically add the original key to the name if desired.
Delete	Removes the gates from the selected Regions and removes them from the list.

Refining the Selection

5. Try out the horizontal and vertical zoom controls in the lower right corner of the waveform display. Zooming in on the wave makes it a lot easier to see exactly what you're doing.
6. Click on one of the numbered region names in the text area. Click on the word "Region" itself, not on the padlocks to the left of the names. The waveform display now automatically zooms in to show a close-up view of the selected region.
7. Press the **Play** button
8. Adjust the **Start** and **End** points of the gates manually so that they conform to the start and end points of the waveform. Just **left-click and drag**, to move them. Feel free to use the zoom controls if you need a closer look.

Manual Settings

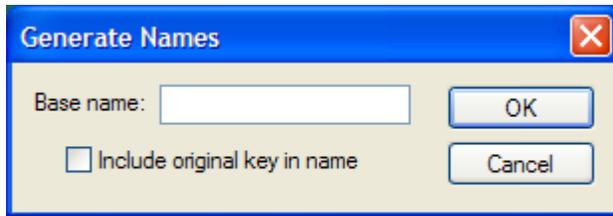
9. The gates settings in the display below have a few problems. The end points have been set too far out and there is an extra Region without a sample.



10. Click on each of the **Region** names one at a time. In the example above, Region 8 is not gating a sample and needs to be deleted. With the Region name selected, click the **Delete** button to discard the gate. Poof!
11. In the example above, the **End** points can simply be dragged inward toward the **Start** point to remove the silence at the end of the sample. Press the **Play** button (or hit the Space bar) to listen to the gated area.

Labeling the Regions

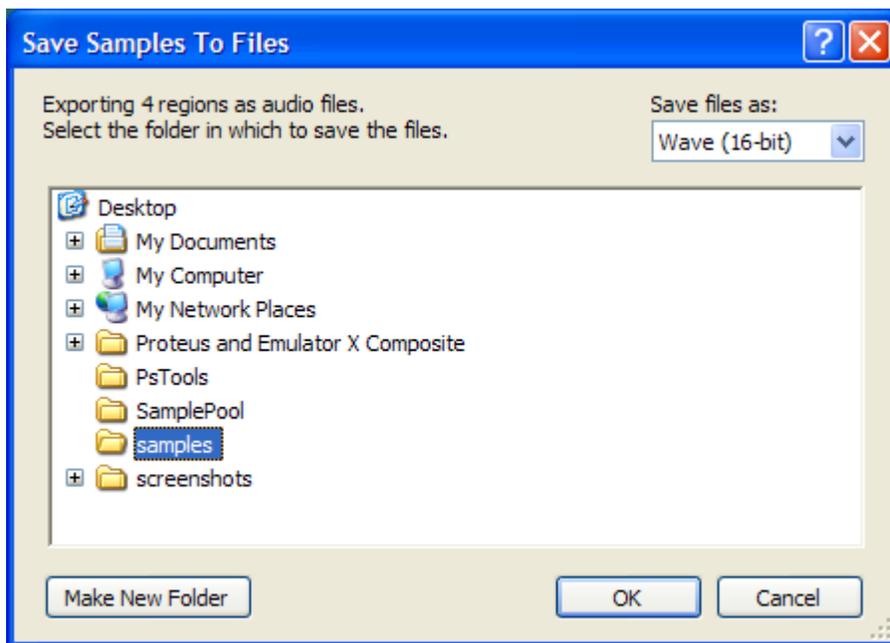
12. Press the **Set Keys...** button in the Edit Section. The Emulator X analyzes each sample and labels it with the correct pitch. These key numbers can be used later for placing the sample on the keyboard.
13. If you know the pitch of the region, you can manually name the regions by clicking over the field and typing in the key number.
14. Click on the **Name** field in the Region List section. The selected section of the waveform is highlighted as you click on the name.
15. The **Set Names...** button allows you to name any selected samples. Shift-click or Ctrl-click to select multiple samples for naming. Use the **Select All** button to choose all regions. You can also include the original key in the sample name to make placement or identification easier.



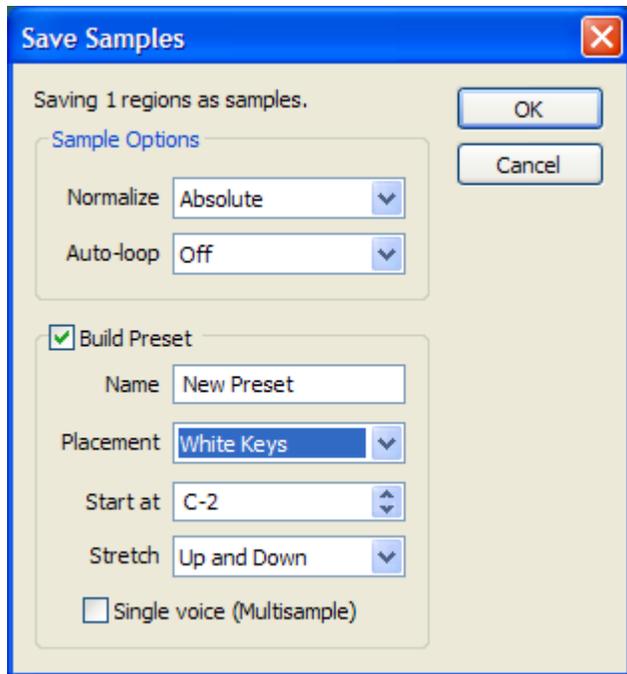
16. Press the headings in the labeling section to **Sort** each column. Press again to sort in reverse order.
17. You can **Lock** each region after you finish setting it up to prevent any accidental changes from being made. Click on the little lock icons to lock and unlock, or use the Lock All or Lock None buttons.

Saving your Samples

1. Once the gates and labels are set, you have several options. You can save the samples as WAVE or AIFF files, add them to the current bank or automatically create a new preset containing the samples.
2. Press the **To Files...** button to save the samples in WAVE or AIFF format. The following pop-up dialog box appears.



3. Choose a location for your samples.
4. From the **Save files as:** drop-down menu, select the file type you want to save as, and click **OK** to save, or **Cancel** to cancel the operation.
5. Press the **To Sampler...** button to add the samples to the current bank. The following pop-up dialog box appears.



6. The save to sampler controls allow you to set up the automatic DSP and placement operations which will normalize your samples, loop them and even create a new preset with the samples placed on the keyboard if you choose.

Save to Sampler Controls

Normalize	Boosts the gain of the samples to make them full-level. Please read “Sample Options” on page 296 for more information.
Auto-loop	This option allows you to choose the algorithm for setting the loop points of the imported samples. Please read “Sample Options” on page 296 for more information.
Build Preset	When checked, this option creates a new preset using your samples. Please read “Build Preset Options” on page 297 for more information.
Name	This Build Preset option allows you to name your new preset.
Placement	This option lets you to specify how samples will be placed in your new preset. Please read “Build Preset Options” on page 297 for more information.
Start At	This option allows you to specify which key will be the first key if you Place the regions on white keys, or chromatically. Please read “Build Preset Options” on page 297 for more information.
Stretch	These options let you control how the samples fill in the empty spaces on keyboard. Please read “Build Preset Options” on page 297 for more information.
Single Voice	If this box is checked, the samples will be placed inside a single Multi-sample voice. See “Multisample Voices” on page 37 for more information.

These features are described in detail in the next section. Please go to [page 296](#) for more information.

7. Click **OK** to perform the Sampler Options and Build the Preset (if enabled). If you chose not to build a preset, the samples will be added to the end of the sample list. If you did build a new preset, a new preset (with your chosen name) will also be added to the end of the preset list in the bank.
8. Open your new preset in the Voice Processing window and modify as desired to complete the preset.
9. Remember to **SAVE THE BANK** or all your work will be lost.

SynthSwipe™

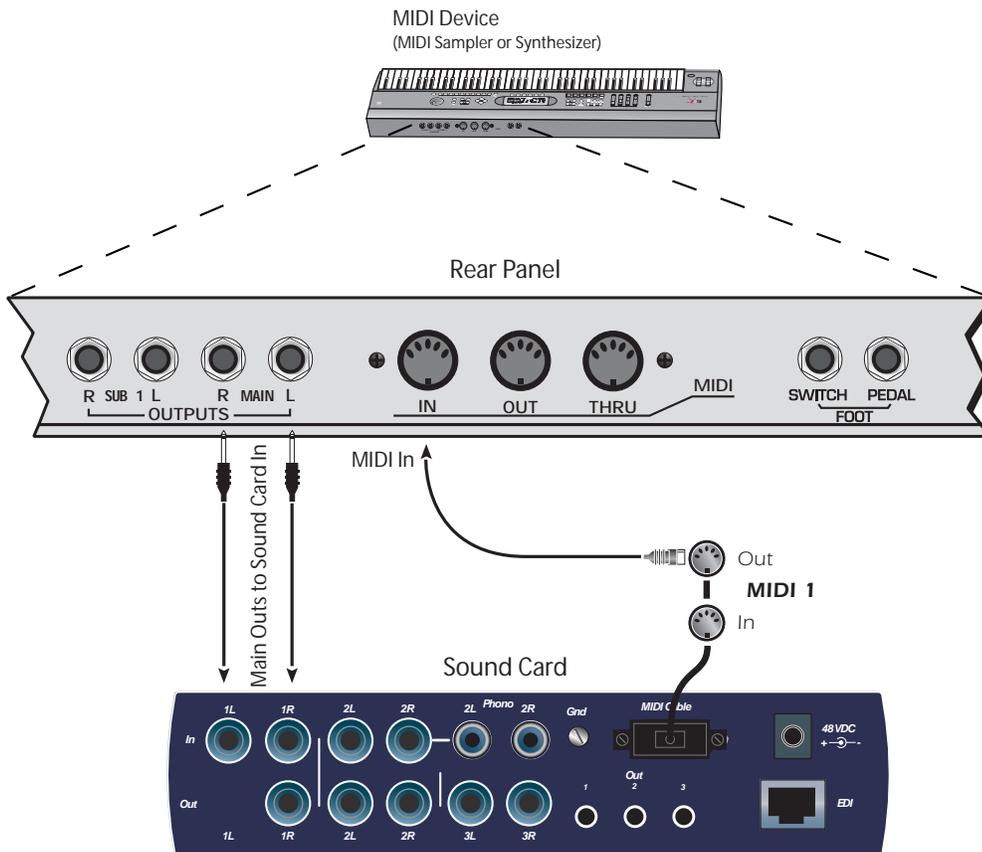
Got an analog instrument with MIDI capabilities, such as a drum machine or a sampler keyboard? You can sample the entire instrument automatically using SynthSwipe.

All you need to do is: Connect your instrument's analog outputs to your computer, connect your computer's MIDI out to your instrument's MIDI in, and run SynthSwipe.

Use the following steps to learn how.

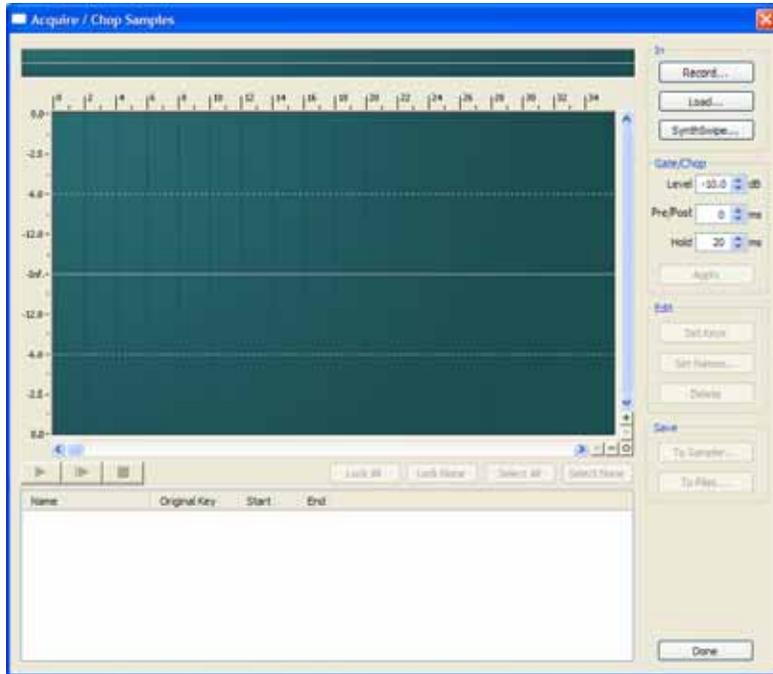
► To Set Up Your Connections

1. Connect the analog line out(s) to your sound card's analog line in(s). Use stereo if possible.
2. Connect the MIDI output on your sound card to the MIDI input on your instrument. See the following diagram:

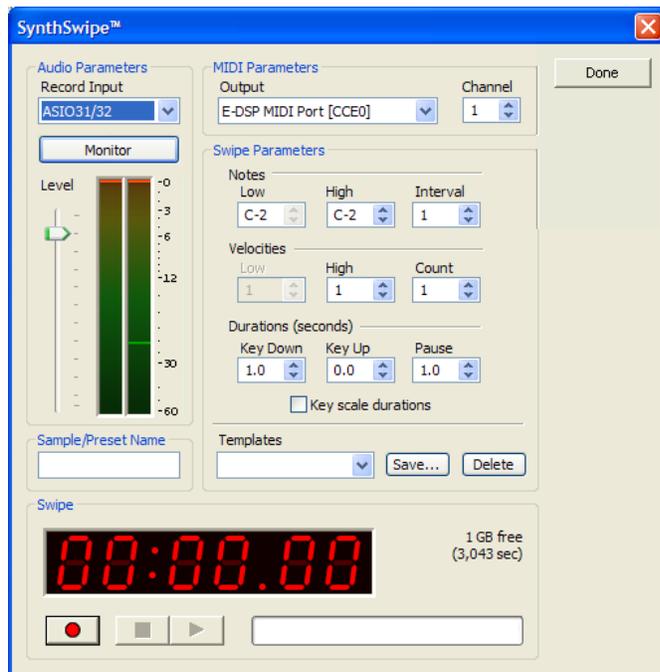


► **To Set Up SynthSwipe**

3. Select **New** from the File menu.
4. From the File menu, select **Acquire Samples...** The Acquire / Chop Samples dialog appears:



5. Click the **SynthSwipe...** button. The SynthSwipe dialog appears.

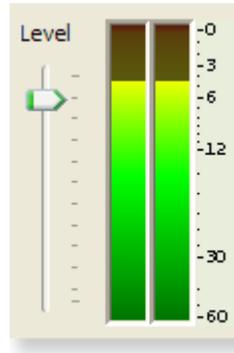


- You can save and load Synthswipe settings as a template. Simply make your settings and click **Save...**The Templates dialog appears. [See “Templates” on page 279.](#)

6. Under **Audio Parameters**, select the analog input(s) that your instrument is plugged into. The displayed input selections are based on which ASIO channels are open when the Record Audio dialog was first opened.

- If no inputs are listed, make sure E-MU ASIO is selected in the Preferences dialog, and that the current PatchMix DSP Session contains at least one ASIO Send.

7. Set your input levels using PatchMix DSP. While playing your instrument, watch your meters. Adjust the input level using PatchMix DSP. The signal peaks should illuminate the yellow area of the meter. The red bar at the very top of the meter indicates clipping and will stay on after clipping is detected. Left-click on either red bar to clear the clip indicator.
8. In the MIDI Parameters section, choose your MIDI output port and channel. The port should be whatever port you're using to connect your computer to your MIDI input on your instrument. The channel must be whatever channel your instrument is currently using.



► **To set the Swipe Parameters**

9. Set the Swipe parameters. The following table explains what each parameter means:

Notes	<p>Low and High - set the range of notes that SynthSwipe will sample. To keep memory and disk usage low, only sample the range that you actually need.</p> <p>Interval - sets the distance in semi tones between sampled notes. For example, setting the low note at C2 and the interval at 7 will sample C's and G's only (G is seven semi tones higher than C).</p>
Velocities	<p>Low and High - set the lowest and highest velocities SynthSwipe will record for each note.</p> <p>Count - sets the number of different velocities at which each note will be played. Note: If you set the count at 1, the Low setting is grayed out and the High setting determines the velocity for each sample. For example, if you set the low at 20, the high at 120, and the count at 3, SynthSwipe would sample each note at a velocity of 20, 70, and 120. You'll probably want to select a Count value between 3 and 5.</p>
Durations	<p>Key Down - sets how long (in seconds) each note is played. SynthSwipe defines a region of the same length.</p> <p>Key Up - sets how long (in seconds) an additional key up region will be recorded after the key down. This is useful if the instrument you are sampling has a tail that sounds after you release the key. SynthSwipe defines a separate region for key up and automatically utilizes this region when you add your newly recorded samples to a bank.</p> <p>Pause - sets how long (in seconds) SynthSwipe waits between the end of the Key Down/Up and the beginning of the next Key Down.</p> <p>Key Scale Durations - decreases the key up time as the pitch of the note increases. This is useful for creating a realistic effect when sampling an instrument that behaves this way, such as an electric piano. A piano's higher notes have shorter sustain times than the lower notes. If you're sampling a piano, checking this box makes the sampled instrument sound more realistic.</p>

10. To begin sampling your instrument, click the **Record**  button. You will hear the recording cycle through all the notes, one velocity at a time. Do not play your instrument during this time.
11. After the recording completes, if you're satisfied with the results, click **Done**. If you want to try again, overwriting what you just recorded, click the **Record** button.
12. You will be prompted to save your bank. Click **OK** and choose a name and location for your new bank.
13. The Save Samples dialog appears. Choose your settings and click OK. [See "Saving your Samples" on page 288.](#)

Your new preset is ready to play with all the Key up, Key down, and velocities automatically set.

Tips for Saving SynthSwipe Recordings

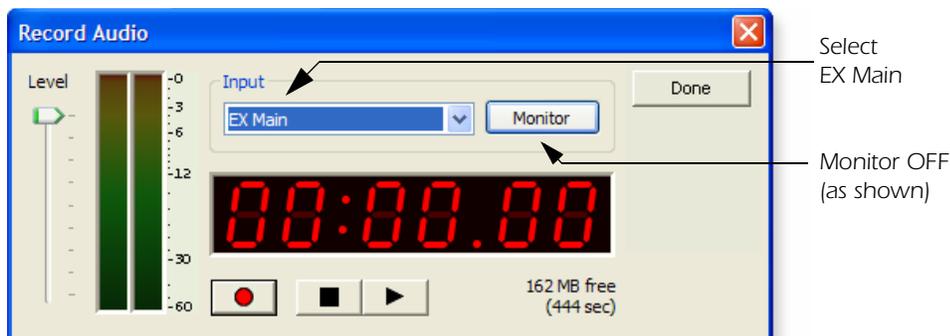
- If you are swiping multiple velocities, turn **Normalize** off in order to preserve the relative volume relationships of the samples.
- For pad-like sounds, choose **Loop 1/2 + Xfade** from the **Auto-Loop** drop-down menu.

Resampling

You can also sample the output of the Emulator X. Resampling allows you to include modulation, filters and effects in the sample, or you can record runs or chords, or a whole song for that matter. The basic idea of resampling is to make the EmulatorX output available as an ASIO input to the Record Audio dialog box. Using PatchMix DSP, this is a simple matter. (Resampling is also possible using other audio interfaces.)

► To Resample the Emulator X using PatchMix DSP

1. Open EmulatorX in standalone mode (not as a VSTi), and select the preset you wish to resample.
2. Open PatchMix DSP by clicking on the EMU icon  in the Systray.
3. Play your MIDI keyboard and determine which input strip Emulator X is using by observing the peak meters. (You can right-click to add peak meters to the channels if there are none.)
4. Add an ASIO Send to the input channel used by the Emulator X.
5. Select **Acquire Samples** from the Emulator X File menu (Ctrl+Q).
6. Press the **Record** button from the "Acquire/Chop Samples" dialog box. The "Record Audio" dialog box appears.



7. Select **EX Main** as the Input. If EX Main doesn't appear, make sure you've loaded the default Emulator X template in PatchMix DSP. Make sure that the Monitor button is off (as shown) or you will hear annoying feedback as soon as you play the Emulator X.
8. **Play your MIDI keyboard.** You should hear it and see activity on the meters in the Record Audio dialog box above.
9. Press the red **Record button**,  then start playing.
10. Press the **Stop button**  when you're finished resampling.
11. Continue to chop and process your samples. See [page 284](#).

Importing Other Sample Formats

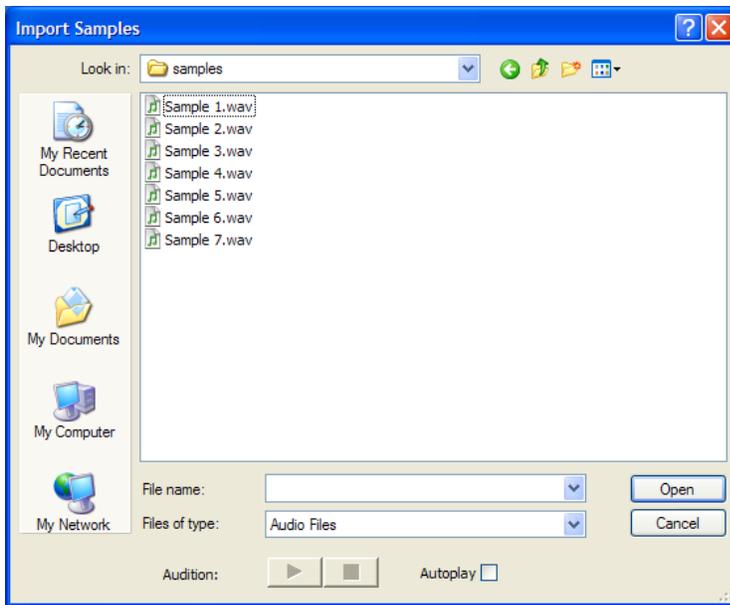
Emulator X can import sample formats such as Wave, AIFF, MP3, and REX 2. Emulator X then automatically builds a preset for you, truncating, looping and placing the samples on the keyboard.

There are three ways to import samples:

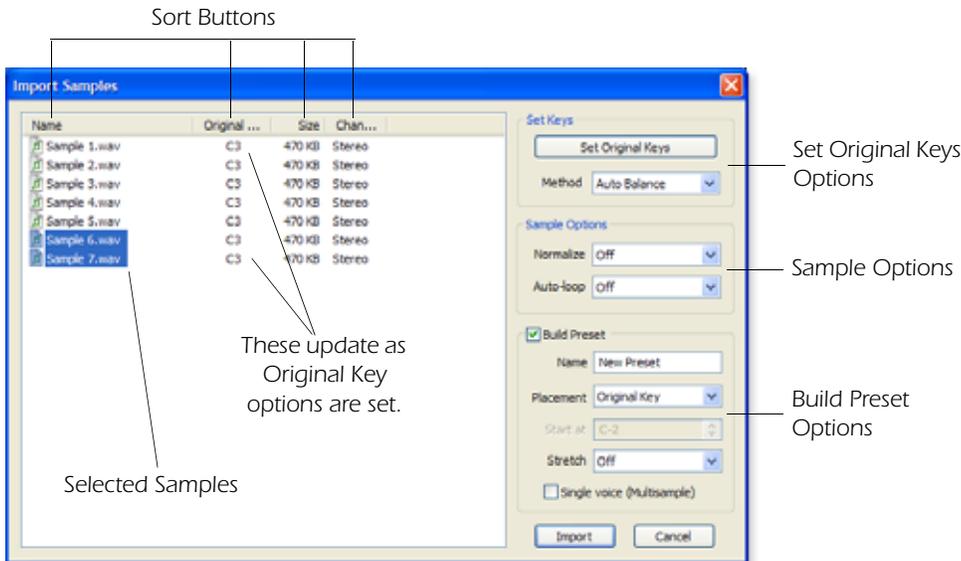
- Select **Import Samples** from the File menu to call up the selection dialog box.
- **Right-Click** with the mouse over the Emulator X icon in the tree, then select "Import Samples" to call up the selection menu.
- **Drag and Drop** the sample files over the Emulator X icon in the tree.

► To Import Samples into the Emulator X

1. When you select "Import Samples", the following dialog box appears.



2. **Locate and select** the desired sample files on your computer, then press **Open**. The following dialog box appears showing the samples you just selected.
 - You can audition a sample by highlighting it and clicking the  button. To stop a sample from playing, click the  button.
 - If you want a sample to audition whenever you click it, check the **Autoplay** box.



There are three groups of controls on the right side of this window.

Set Original Keys	Options which determine the pitch of each sample.
Sample Options	Automatic sample trimming and looping options.
Build Preset Options	These options determine the keyboard placement of each sample.

Set Original Keys

The original key of a sample determines its pitch on the keyboard so it's important that it be correctly placed. This group of controls allows you to select an automatic placement method for the currently selected samples.

The **Set Keys** button allows you to immediately process a key setting algorithm for a set of samples. The Original Key field in the sample area changes to show the new original key. You can then choose another algorithm for other samples. This allows you to try out the algorithms and set each sample to the proper original key before importing.

Original Key Algorithms

Embedded Info	Keys are set according to key data embedded within the sample files themselves, if any.
File Name	Keys are set by attempting to extract key names from the sample file names. (A#3, B2, etc.)
Pitch Detect	Keys are set by algorithmically measuring the fundamental pitch of the sample file.
Auto Balance	Keys are set by dividing the available key range equally among the selected sample files.
White Keys	Keys are set on successive white keys beginning with the lowest key in the 88-key keyboard range (A-1).
Chromatic	Samples will be placed within the preset on successive keys, beginning with the lowest key in the 88-key keyboard range (A-1).

Sample Options

These sample options are applied to all selected samples.

The **Normalize** option allows you to specify how amplitude normalization should be applied to stereo samples.

Absolute	Each side of the stereo sample is individually normalized without regard to the other channel.
Relative	Each side of the stereo sample normalized by the same gain factor, so that the two channels maintain their relative volume.
Off	No normalization processing will be performed.

The **Auto-Loop** option allows you to choose the algorithm for setting the loop points of the imported samples. Auto Looping attempts to find the best loop point using auto-correlation (See [page 178](#)). The auto-looping options are listed below.

◆ For additional information, see [Normalize on page 187](#).

Auto-loop Algorithms

Whole	The loop points on each imported sample are set to enclose the entire sample.
1/2	The loop points on each imported sample are set to enclose the last half of the sample.
1/2 + Crossfade	The loop points on each imported sample are set to enclose the last half of the sample, including an appended crossfade to the loop start.
1/3	The loop points enclose the last 1/3 of the sample.
1/3 + Crossfade	The loop points enclose the last 1/3 of the sample, including an appended crossfade to the loop start.
1/4	The loop points enclose the last 1/4 of the sample.
1/4 + Crossfade	The loop points enclose the last 1/4 of the sample, including an appended crossfade to the loop start.
1/8	The loop points enclose the last 1/8 of the sample.
1/8 + Crossfade	The loop points enclose the last 1/8 of the sample, including an appended crossfade to the loop start.

Tips for using Auto Loop

- Use longer loops (1/2) and crossfade on harmonically complex samples such as piano and strings.
- Short loops work well on simpler waveforms such as brass, flutes, guitars, etc.
- **Don't** choose crossfade if you plan to fine tune your sample loops later.

▼ **Warning:**
Crossfading permanently modifies the sample data. For additional information, see [page 204](#).

Build Preset Options

These sample options allow you to specify if a new preset should be built using the imported samples.

When **Build Preset** is checked, a preset is created during import using all of the imported samples. Samples will be ordered within this preset identically to their ordering in the sample file list. The preset will be automatically named according to the first imported sample.

If “**Build Preset**” is unchecked, a preset will not be created during import, and the imported samples will simply be added to the instrument’s sample pool. The other controls in this group will be disabled (greyed-out) if “**Build Preset**” is unchecked.

Multisample Check Box - You have the option to build multiple voices (with one sample each) or a single multisample voice. [See “Multisample Voices” on page 37.](#) Check the checkbox to build one multisample voice.

The **Placement** option allows you to specify how samples will be placed in the constructed preset. (Note that this is a separate setting from the “**Original Key**” assignment. The former setting is a property of each sample and denotes where the sample may be placed so that it plays back at unity pitch. This setting relates to the actual per-sample assigned key range within a preset.) The Placement options are listed below.

Sample Placement Algorithms

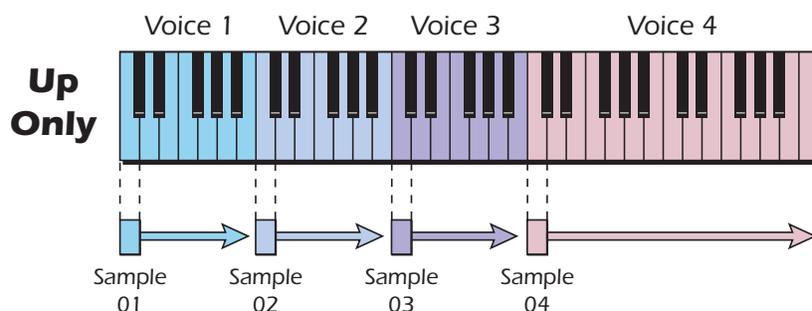
Original Key	Samples will be placed within the preset according to their original key settings.
White Keys	Samples will be placed within the preset on successive white keys, beginning with the key specified in the Start at drop-down menu.
Chromatic	Samples will be placed within the preset on successive keys, beginning with the key specified in the Start at drop-down menu.

The **Start at** options let you choose the first note for placement if you choose **White Keys** or **Chromatic**.

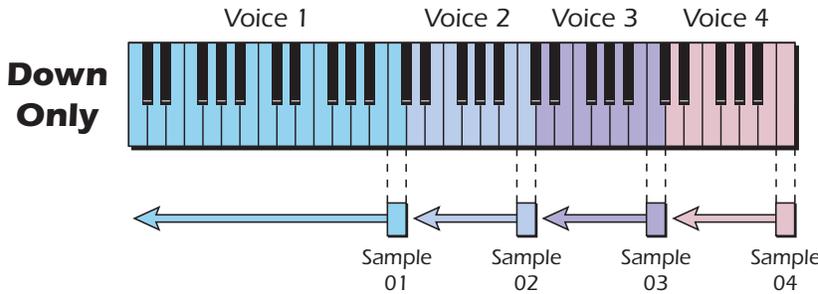
The **Stretch** options let you control how the samples fill in the empty spaces on the keyboard. If you’re only importing four samples, C1, C2, C3, and C4, you’ll probably want to stretch the voice range in order to fill in the missing keys. The Stretch options are listed below.

Up Only

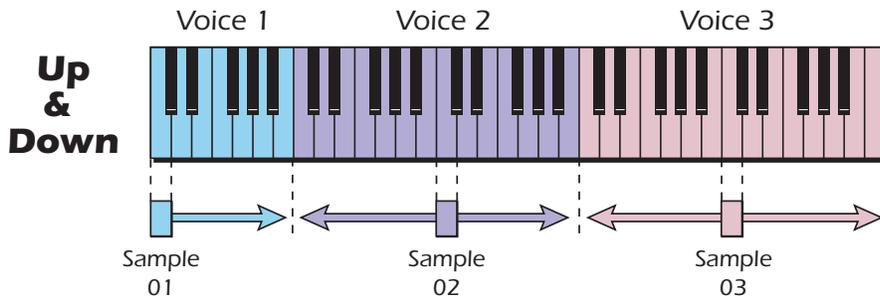
The assigned key range of each imported sample extends upward from the specified placement to the key immediately below the placement of the next imported sample. The key range of the last imported sample extends up to the highest note in the 88-key keyboard range (C7).



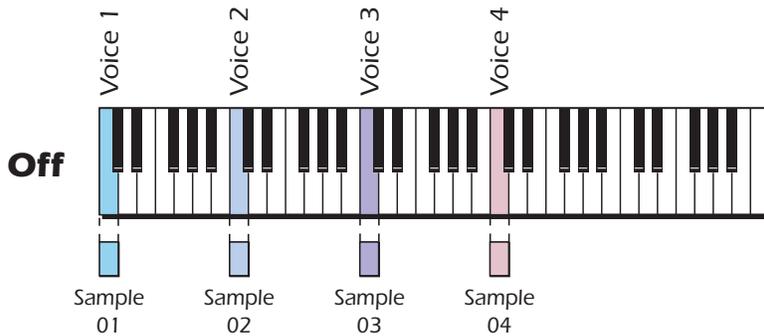
Down Only The assigned key range of each imported sample extends downward from the specified placement to the key immediately above the placement of the previous imported sample. The key range of the first imported sample extends down to the lowest note in the 88-key keyboard range (A-1.)



Up & Down The assigned key range of each imported sample extends upward and downward from the specified placement, such that the available key range between imported samples is split evenly between them. If there are an uneven number of keys between placements of adjacent samples, the contested key will be assigned to the upper key range. The key range of the first imported sample shall extend down to the lowest note in the 88-key keyboard range (A-1), and the key range of the last imported sample shall extend up to the highest note in the 88-key keyboard range (C7.)



Off Each imported sample is only assigned to the placement key.



12 - Synthesizer Basics

This chapter contains background information on the various programming aspects of synthesizers.

If you are new to synthesizers and electronic music, you may need more background information than this manual provides. There are many books dedicated to synthesizer basics and MIDI available through your local music dealer. Magazines such as Keyboard and Electronic Musician, available at most newsstands, contain current information on the subject, as well as valuable programming tips. The internet is also a rich source of information on this subject. Seek and you shall find.

Your initial involvement with Emulator X will most likely consist of using the existing banks and presets. While our factory banks and presets are very good, there are probably some things you would like to change, perhaps the LFO speed, the filter setting, or the attack time. Eventually, you'll want to make your own custom presets using complex modulation routings.

Emulator X Advanced Applications Guide

The Emulator X Advanced Applications Guide is another valuable resource which contains 43 more in-depth tutorials. The Advanced Application Guide begins with the fundamentals and walks you through nearly every facet of this amazing instrument. You can find this pdf document on your Emulator X3 CD or online at: www.emu.com.

Editing Presets

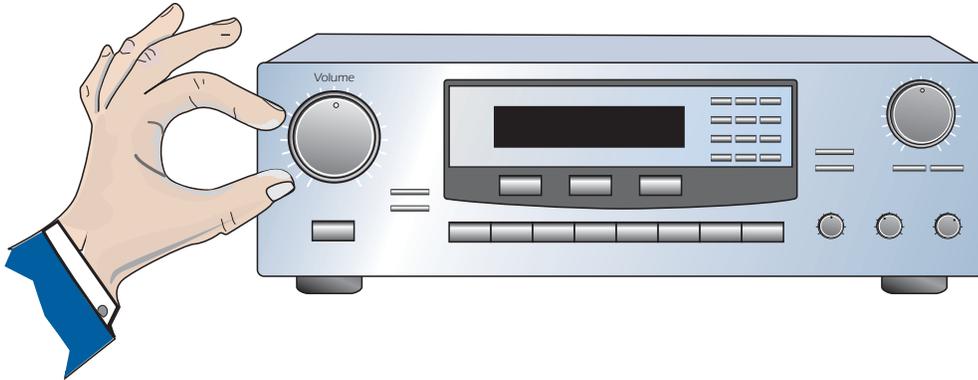
It's easy to create new presets by modifying existing presets. This is really the best way of getting acquainted with Emulator X. If you don't like the results, simply reload the preset or bank and you'll be back to the original sound. Changes are not made permanent until you SAVE a bank. Therefore, you can experiment all you want with presets, voices and samples without worrying about losing a sound.

We encourage you to actually try out the different functions as you read about them. Hearing what a control actually does will remove a lot of the mystery associated with it.

Emulator X contains an extensive modulation implementation using two multi-wave LFO's (Low Frequency Oscillators), three envelope generators, and the ability to respond to multiple MIDI performance controllers. You can simultaneously route any combination of these control sources to multiple destinations.

Modulation

Modulation means to dynamically change a parameter, whether it be the volume (amplitude modulation), the pitch (frequency modulation), or whatever. Turning the volume control on your home stereo rapidly back and forth would be an example of amplitude modulation. To modulate something we need a modulation source and a modulation destination. The source is your hand turning the knob, and the destination is the volume control. If we had a device that would automatically turn the volume control, we would also call that device a modulation source.



Turning the volume control back and forth on your home stereo is an example of Amplitude Modulation.

Emulator X is designed so that for each of the variable parameters, such as the volume, there is an initial setting which can be changed by a modulation source. Therefore in the case of volume, we have an initial volume and we can change or modulate that volume with a modulation source. Positive modulation *Adds* to the initial amount. Negative modulation *Subtracts* from the initial amount.

Modulation Sources

The main modulation sources are Envelope Generators, Performance Controllers and Low Frequency Oscillators. In the previous example, an envelope generator could be routed to automatically turn the volume control as programmed by the envelope, or, a low frequency oscillator could be routed to turn the volume control up and down in a repeating fashion. The main mod sources are listed below.

Keyboard Key

Which key is pressed.

Key Velocity

How fast the key is pressed.

Release Velocity

How fast the key is released.

Gate

High if the key is pressed, low when the key is released.

Key Glide

A smoothly changing control source based on the Glide Rate and the interval between the last two notes played.

Pitch and Mod Wheels

Keyboard pitch bend and modulation wheels.

Keyboard Pressure (mono aftertouch)

Key Pressure applied after the key is initially pressed.

Pedal

A continuously variable pedal controller connected to your MIDI keyboard.

Miscellaneous Controllers A-P

Any type of MIDI continuous controller data from your keyboard or other controller. MIDI continuous controller numbers are programmed in the Preferences dialog box (See [page 28](#)).

Low Frequency Oscillators (2 per voice)

Generate moving, repeating waves used to add interest and animation to the sound.

Envelope Generators (3 per voice)

Generate a programmable “contour” which changes over time when a key is pressed.

Function Generators (3 per voice)

A programmable, stepped modulation source which can be used as a complex LFO, a sequencer or an arpeggiator depending on how it is utilized.

Noise & Random Generators

Generate spectrums of noise and random signals to be used for modulation.

T-switch and Footswitches

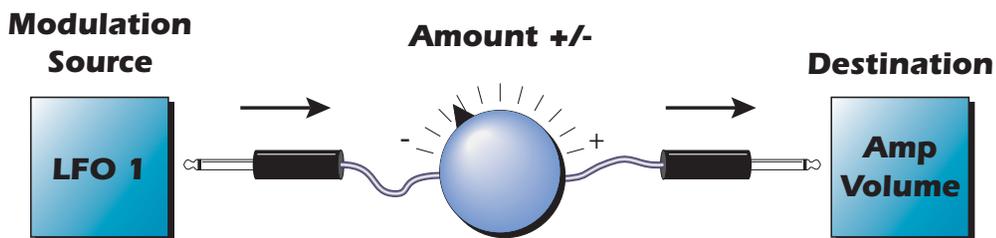
Change a parameter when a MIDI switch is pressed. MIDI footswitch numbers are programmed in the Preferences dialog box (See [page 28](#)).

Modulation Cords

You use a Cord to connect a modulation Source to a Destination. (Patchcord – a holdover from the days when modules of an analog synthesizer were connected together with physical cords. Nowadays, we still need a way to connect modules together, but the cords are in software.)

You can connect the modulation sources in almost any possible way to the modulation destinations. You can even modulate other modulators. Each cord also has an amount parameter which determines “how much” modulation is applied to the destination. The modulation amount can be positive or negative and will either add or subtract from the initial value.

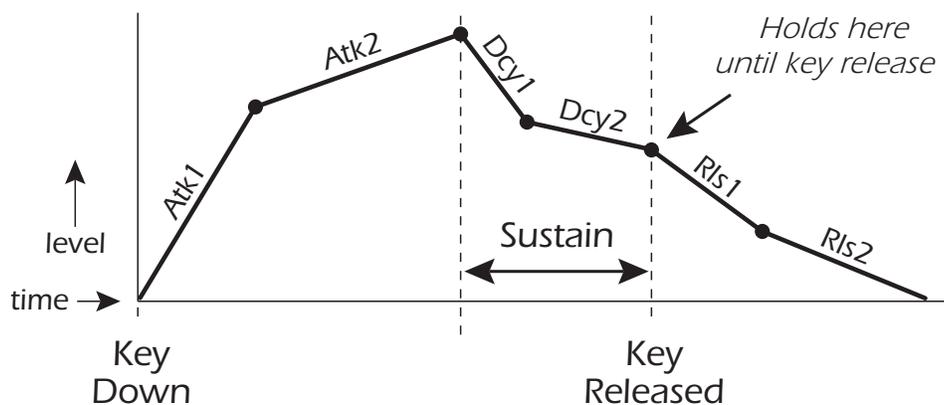
To use a modulation cord, you must connect a modulation Source to a modulation Destination. Think of modulation cords as if you are connecting an actual cord. You must connect both ends of the cord for the connection to be made. There are 36 general purpose cords per voice.



Envelope Generators

An envelope can be described as a “contour” which can be used to shape the sound in some way over time. There are three envelope generators per voice and all of them are the rate/level type.

This is how the rate/level envelopes work: When a key is pressed, envelope starts from zero and moves toward the Attack 1 Level at the Attack 1 Rate. As soon as it reaches this first stage, it immediately begins the Attack 2 phase and moves toward the Attack 2 level at the Attack 2 rate. As long as the key is still held, the envelope continues on through the Decay 1 and Decay 2 stages. If the key is still held when the envelope reaches the end of Decay 2, it simply stops there waiting for you to release the key. When you release the key, the envelope continues through its Release 1 and Release 2 stages, stopping at the end of the Release 2 stage. The rate/level envelopes give maximum flexibility to program both complex and simple envelopes.



The Amplifier Envelope generator controls the volume of the voice over time and has 6 stages: Attack 1, Attack 2, Decay 1, Decay 2, Release 1 and Release 2. The Filter Envelope generator controls the filter morph and also has 6 stages. Unlike the amplifier envelope, however, the filter envelope has the ability to go negative as well as positive. There is also an Auxiliary Envelope generator which is a general purpose envelope. The auxiliary envelope is identical to the filter envelope and can go negative as well as positive. The time of each stage can be adjusted to create myriad envelope shapes, which in turn shape the sound over time.

The way the volume of a sound changes over time determines how we perceive that sound. For example, a bell struck with a hammer is instantly at full volume, then slowly dies away. A bowed violin sound fades in more slowly and dies away slowly. Using the Amplifier Envelope, you can simulate different types of instrument volume envelopes by programming them appropriately.

Retriggering

The Filter and Auxiliary envelopes can be retriggered by an LFO or any other modulation source such as clock divisor. (See “Clock Modulation” on page 304.) Patch the desired modulation source (using a positive value) to Filter Env Trigger or Aux Env Trigger. Envelopes are triggered on a positive-going edge.

Complete Mode

In Complete mode, the envelope generator continues through all six stages when a key is pressed, ignoring the status of key sustain. This feature allows you to create long evolving soundscapes with a quick tap of the key, enveloping percussive sounds such as piano and drums, or plucked strings. This mode is also useful to conserve CPU resources, since the voice will turn off as soon as the amp envelope cycle completes.

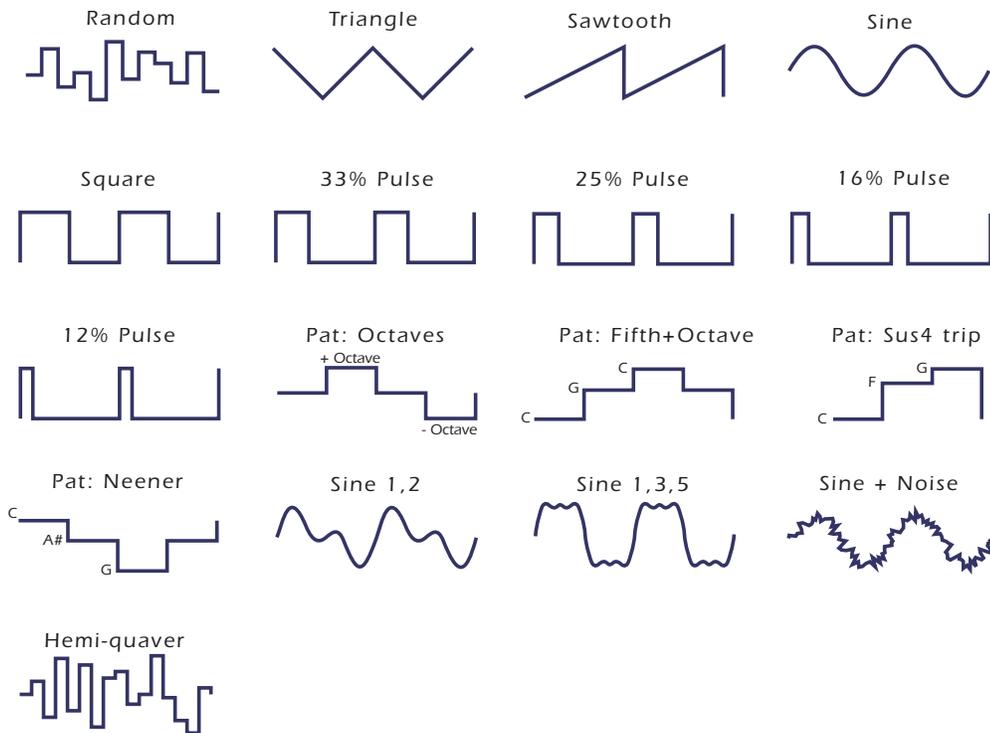
■ The EOS envelope generators are just like standard ADSRs, except that they have two segments for each stage.

To create a standard ADSR curve, set the “2” levels the same as the “1” levels and set all the “2” rates to 0.

■ By routing the Auxiliary Envelope to control the pitch (Cords) you can easily hear the shape of the envelopes you are creating.

Low Frequency Oscillators (LFOs)

A Low Frequency Oscillator or LFO is simply a wave which repeats at a slow rate. Emulator has two multi-wave LFOs for each channel. The LFO waveforms are shown in the following illustration.



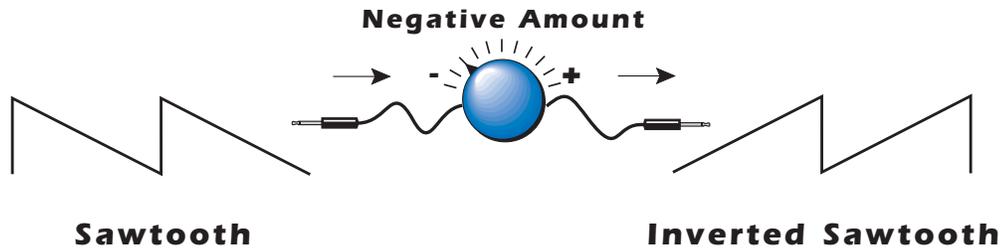
By examining the diagram of the LFO waveforms, you can see how the LFO will affect a modulation destination. Suppose we are modulating the pitch of an instrument. The sine wave looks smooth, and will smoothly change the pitch. The square wave changes abruptly, and will abruptly change the pitch from one pitch to another. The sawtooth wave smoothly decreases, then abruptly changes back up. The sound's pitch will follow the same course. Controlling the pitch of an instrument is an easy way to hear the effects of the LFO waves.

Like the Auxiliary Envelope, the LFOs can be routed to control any realtime functions such as Pitch, Filter, Panning, or Volume. A common use for the LFO is to control the pitch of the sound (LFO -> Pitch). This effect is called vibrato and is an important performance parameter. Many presets use this routing with the modulation wheel controlling "how much" LFO modulation is applied. Another common effect, Tremolo, is created by controlling the volume of a sound with the LFO (LFO -> Amp Volume).

Another use for the LFOs might be to add a slight bit of animation to the sound by routing the LFO to control the filter. In this example, the Cord amount would be set low, for a subtle effect.

When the amount of an LFO is a negative value, the LFO shape will be inverted. For example, inverting the sawtooth wave produces a wave that smoothly increases, then instantly resets down. Since the inverted sawtooth wave is now negative, adding DC with a cord amount of +100 will make it positive again. See [page 308](#) for information on how to use a DC Cord.

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



Random Sources

Random modulation sources can be used when you want the timbre of the sound to be “animated” in a random or non-consistent manner.

Key Random 1 & 2 generate different random values for each voice which are selected at key-on time and do not change during the note.

The White & Pink Noise Generators produce varying random values. Both white and pink noise sources are low frequency noise designed for control purposes. Either noise source can be filtered even more by passing it through a lag processor.

The Crossfade Random function generates the same random value for all voices in a preset. This source is designed to be used for randomly crossfading voices, although you may find other uses.

The Function Generators have a random mode. This can be used to randomly pick from a set of pre-programmed values, such as the notes in a particular key.

Clock Modulation

The clock can also be used as a modulation source. It can be used to trigger the Filter or Auxiliary envelope generators, trigger sample start, synchronize the LFOs, or used directly as a square wave modulation source. Envelopes are triggered on the positive going edge of the clock. LFOs are triggered on the negative going edge.

The Clock source is available in six divisions (double whole note, whole note, half note, quarter note, eighth note, sixteenth note). The different rates can be used separately or in conjunction to create complex rhythm patterns.

Using MIDI clocks, the internal clock can also be synchronized with an external MIDI device such as a drum machine or sequencer.

Clocks are routed exactly like the other modulations sources using the Cords. The Cord Amount MUST be positive (+) for the clock to pass. By modulating the Cord Amount, the divided clocks can be routed around using realtime controllers or other modulation sources.

Syncing an LFO to the Clock

When an LFO is triggered by a clock, the LFO wave resets to zero every time the clock wave goes low. To sync an LFO to the clock, patch a clock divisor to the LFO Trigger (Trg) in the Cords screen.

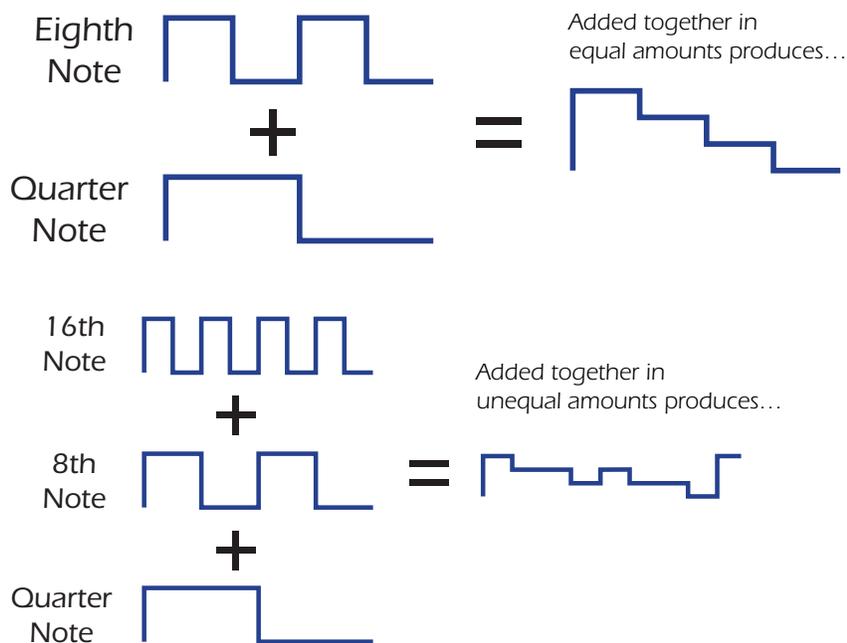
If the LFO rate is close to the clock rate, **the LFO will synchronize with the clock**. If the two rates are far apart, the waveform of the LFO will be mildly or radically altered (as shown by the diagram on the following page).

◆ Envelopes are triggered on the positive going edge of the clock. LFOs are triggered on the negative going edge of the clock.

◆ The tempo of the master clock is set from the Multisetup page.

The possibilities of clock modulation and retrigger are numerous. A repeating six segment curve of any shape can be created by triggering the Filter or Auxiliary Envelope generators with the clock. A few other possibilities are listed below.

- Turn different voice layers on and off using different clock divisors.
- Route the Clock through a Lag Processor to smooth the sharp edges or to create a global triangle LFO.
- Switch between Auxiliary and Filter Envelope retriggering using a slider or footswitch.
- Retrigger LFOs or Envelopes using noise or other LFOs to create random or semi random effects.
- Modulating the rate of triggered LFOs will alter the LFO waveform.
- Route multiple clocks with different divisors to the same destination (such as pitch) to create complex patterns. (Hint: Adjust the Cord Amounts.)



Function Generators

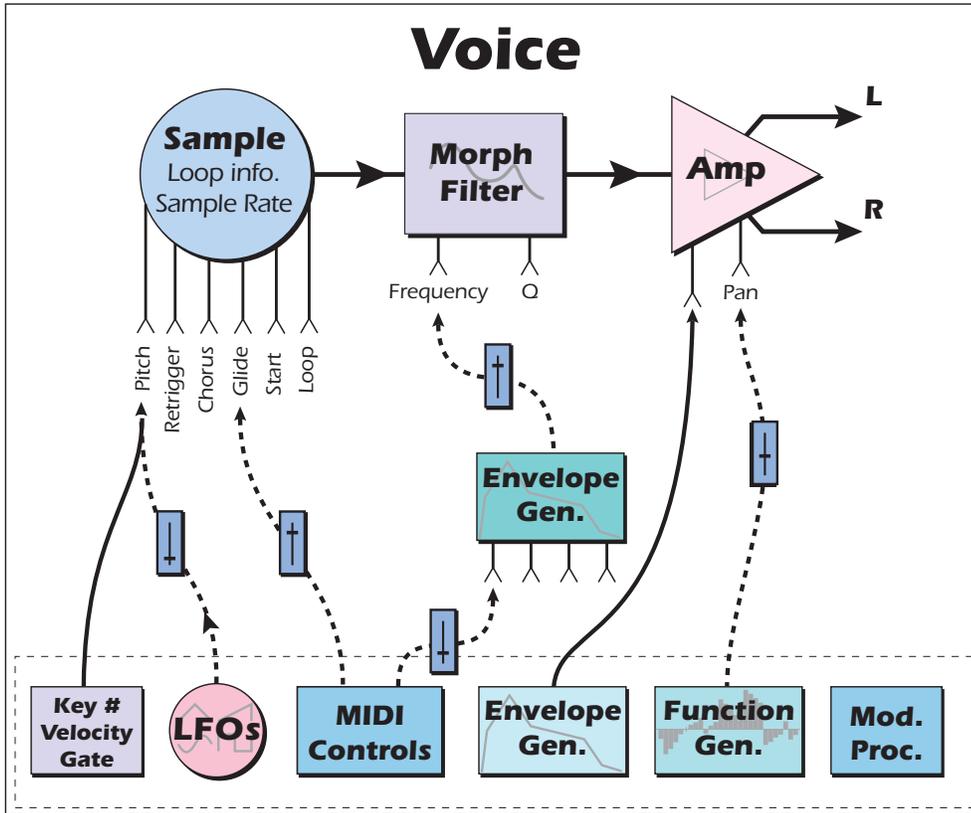
The function generators are multi-function modulation sources and can be used as complex LFOs, sequencers, arpeggiators, or envelope generators. Each voice has three independent function generators. The function generator outputs a stepped (or smooth) sequence of values and a gate/trigger for each step.

Its most obvious function is as a step-sequencer/arpeggiator (when controlling pitch), or as a complex event generator for the filter or amplifier. Several key sync and direction modes allow you to create interesting variations and complex levels of control. The gate and trigger outputs can trigger envelope generators, retrigger samples or LFOs, accent certain notes, change tempo, and trigger other events.

Please see ["Function Generators" on page 139](#) for detailed information and examples of how to use this versatile modulation source.

Modulation Destinations

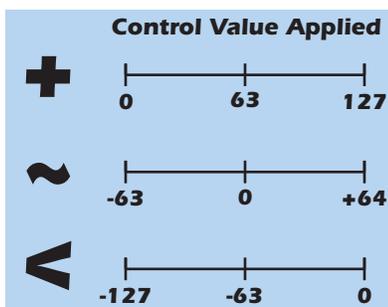
The Cords section of the Preset Edit module is where you connect sources to destinations. Each cord has an amount control associated with it to control how much modulation is applied.



The Cords screen and the diagram above show how modulation sources are connected to destinations. The modulation sources can control any of the destinations in the voice.

Note-on modulation sources, such as key, velocity and gate output a single value at note-on time. Realtime modulation sources such as LFOs, envelope generators and modulation wheels can be varied continuously.

The possible modulation routings are completely flexible as shown in the diagram above. Multiple sources can control the same destination, or a single source can control multiple destinations.



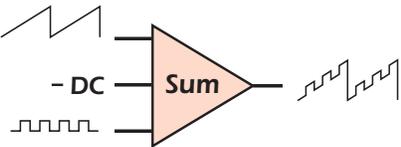
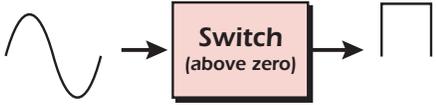
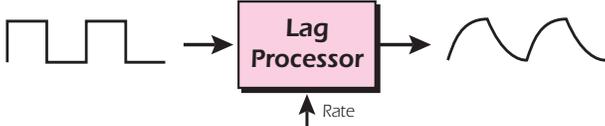
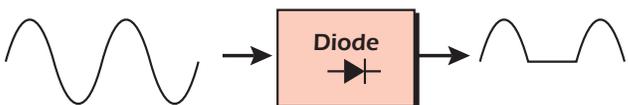
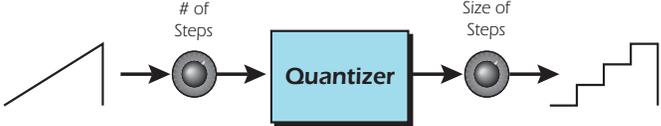
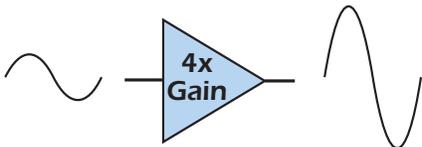
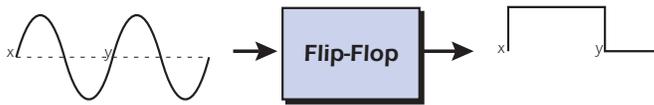
+ modulation ADDS to the initial value. (Normal)

~ centers around Zero. (Use for LFOs, Filt. Freq.)

< modulation SUBTRACTS from the initial value

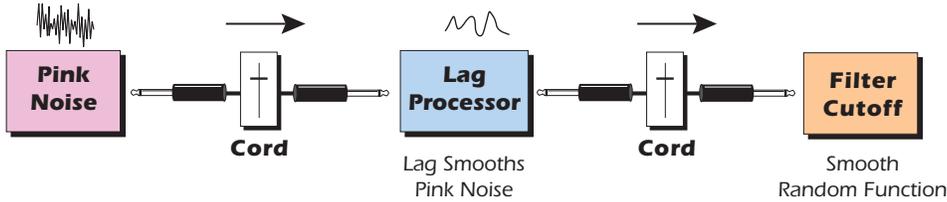
Modulation Processors

Modulation processors are devices which can modify modulation sources such as LFOs and envelope generators before they are applied to a destination. The modulation processors allow you to create patches and do tricks which would not be possible otherwise. The following modulation processors are currently implemented:

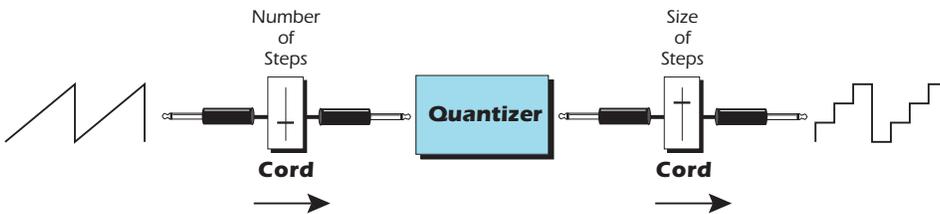
<p>Summing Amp Allows you to add several modulation sources so the output can be controlled by a single Cord Amount.</p>	
<p>Switch Outputs a full value (100) level whenever the input is greater than zero.</p>	
<p>Lag Processors Slow down rapid changes in the input signal. The output "lags" behind the input. Rate is adjustable.</p>	
<p>Absolute Value Inverts negative values and leaves positive values untouched. Outputs only positive values.</p>	
<p>Diode Blocks negative values and allows positive values to pass untouched.</p>	
<p>Quantizer With the input Cord set to 100% the output is limited to 16 discrete steps.</p>	
<p>4x Gain This processor multiplies the input value by a factor of four. Used to amplify modulation sources.</p>	
<p>Flip-Flop Alternates between 1 and 0 each time the input goes positive from a negative input.</p>	

Modulation Processor Examples

This example routes the Pink Noise generator through one of the Lag Processors in order to derive a smooth random function. A smooth random wave is useful in small amounts to add a degree of natural variation to timbre when routed to filter cutoff. Normal pink noise is low pass filtered audio frequency noise with a 3 dB/octave slope to give equal energy per octave. Our pink noise is actually more like very low frequency filtered (mauve?) noise, but it is perfect for use as a random control source.



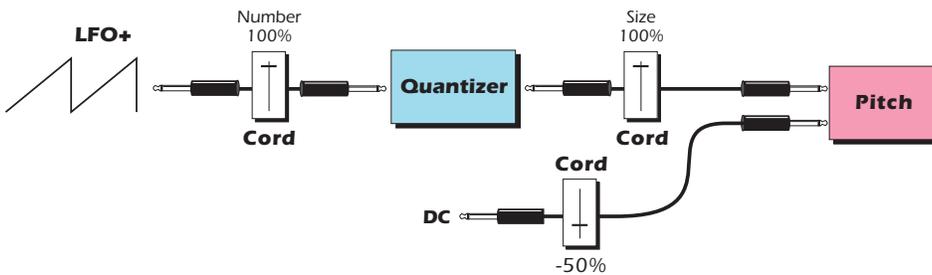
The Quantizer can generate interesting whole-tone scales when envelope generators or LFOs are routed to the input. The quantizer turns a smoothly changing input signal into a series of steps. By routing the output of the quantizer to Pitch and adjusting the cord amounts, you can control both the number of steps and the pitch interval of each step.



The input cord amount limits the number of steps. A sawtooth wave (LFO+) feeding the input, and the cord amount set to 100% generates sixteen steps. The output cord amount controls the size (or interval) of the steps.

This patch generates an ascending arpeggio every time you press a key. The block diagram of the patch is shown below. The patch is very straightforward except for the DC offset which was added in to bring the pitch down into tune. (Sometimes you have to fix a problem, but thanks to the mod processors there's usually a way around it to achieve the desired result.)

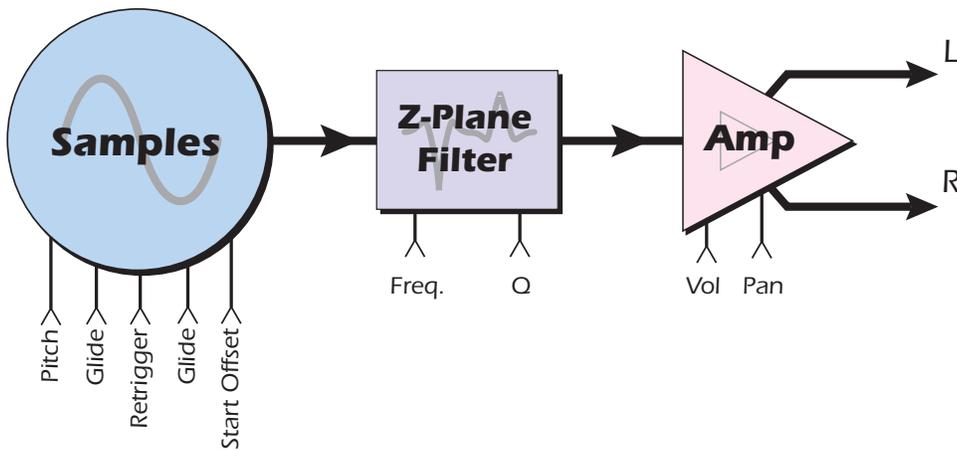
■ Experiment with this patch by connecting other sources and destinations to the quantizer.



You can probably start to see some of the possibilities (and there are many). Whenever you find yourself wishing for some esoteric type of control, take a minute and think if there is a way to achieve the desired result using the modulation processors.

Dynamic Filters

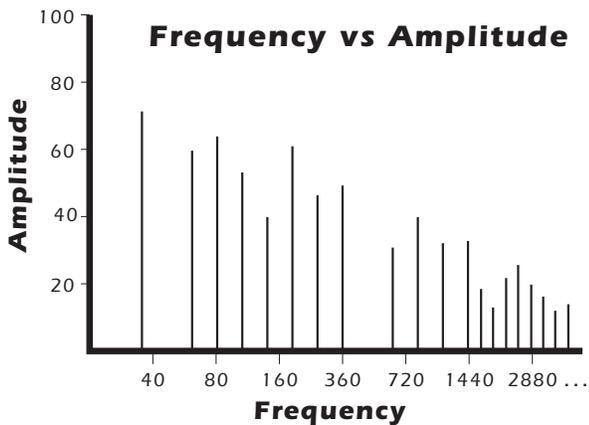
Following is a block diagram of the signal path.



Emulator X utilizes advanced Z-plane filters which can dramatically alter the sound of a sample.

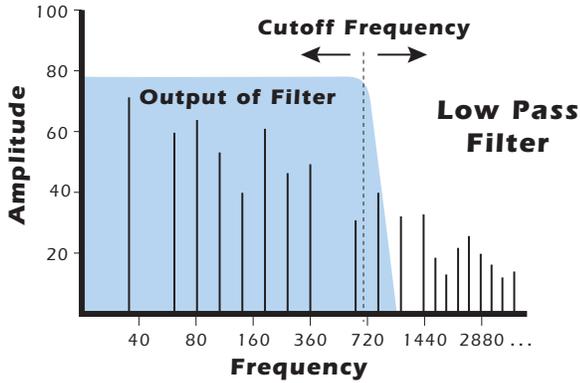
To understand how a filter works, we need to understand what makes a sound wave. A sine wave is the simplest form of sound wave. Any waveform except a sine wave can be analyzed as a mixture of sine waves.

One way to represent complex waveforms is to use a chart with frequency on one axis and amplitude on the other. Each vertical line of the chart represents one sine wave at a specific amplitude and frequency.

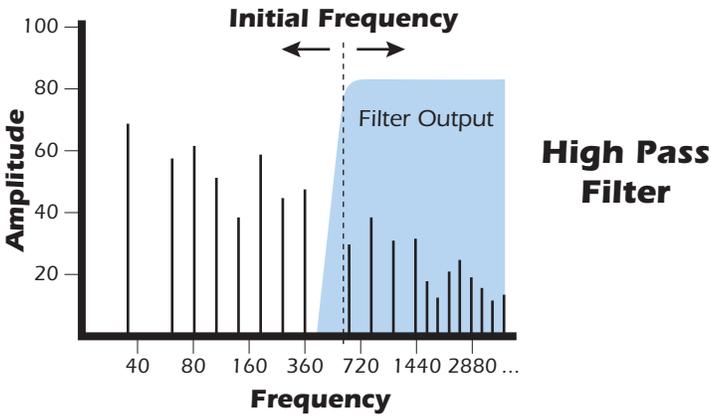


What is a Filter?

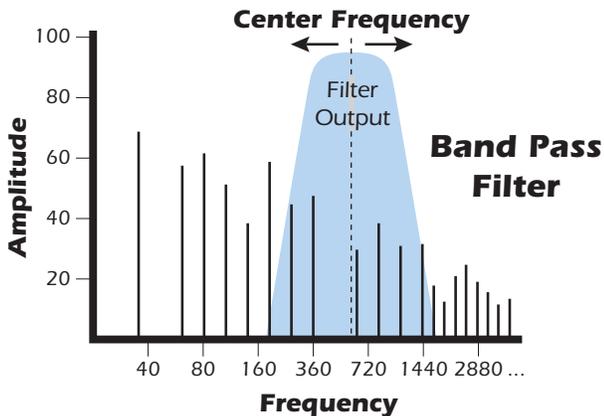
Most samples are complex waves containing many sine waves of various amplitudes and frequencies. A filter is a device which allows us to remove certain components of a sound depending on its frequency. For example, a Low Pass Filter lets the *low frequencies pass* and removes only the high frequencies.



A filter that lets only the high frequencies pass is called a high-pass filter.

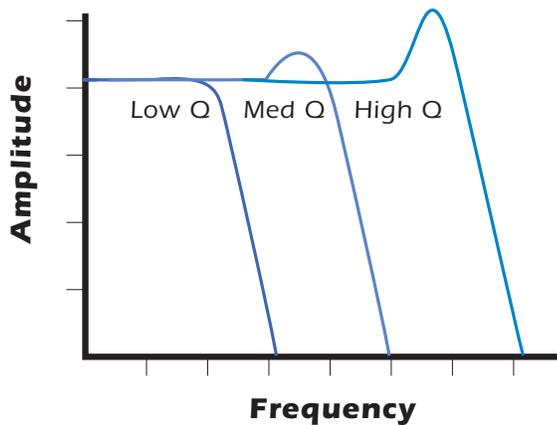


A filter that only lets a certain band of frequencies pass is called a band-pass filter.

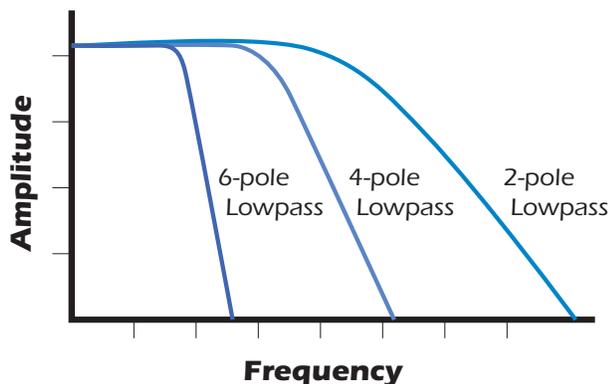


A Notch Filter is just the opposite of a bandpass filter and is used to eliminate a narrow band of frequencies.

Another control found on traditional filters is called Q or resonance. A lowpass filter with a high Q would emphasize the frequencies around the cutoff frequency. The following chart shows how different amounts of Q affect the low pass filter response. In terms of sound, frequencies around the cutoff tend to “ring” with high Q settings. If the filter is slowly swept back and forth, with a high Q , various overtones are “picked out” of the sound and amplified as the resonant peak sweeps over them. Bells and gongs are real world examples of sounds which have a high Q .



Another characteristic of a filter is the number of poles it contains. Traditional synthesizer filters were usually either 2-pole or 4-pole filters. Emulator X has a selectable 2, 4 or 6-pole low pass filter. The number of poles in a filter describes the steepness of its slope. The more poles, the steeper the filter's slope and the stronger the filtering action. The tone controls on your home stereo are probably one-pole or two-pole filters. Parametric equalizers are usually either two-pole or three-pole filters. In terms of vintage synthesizers, Moog and ARP synthesizers used 4-pole filters. Oberheim and E-mu synthesizers were famous for their 2-pole filter sound.



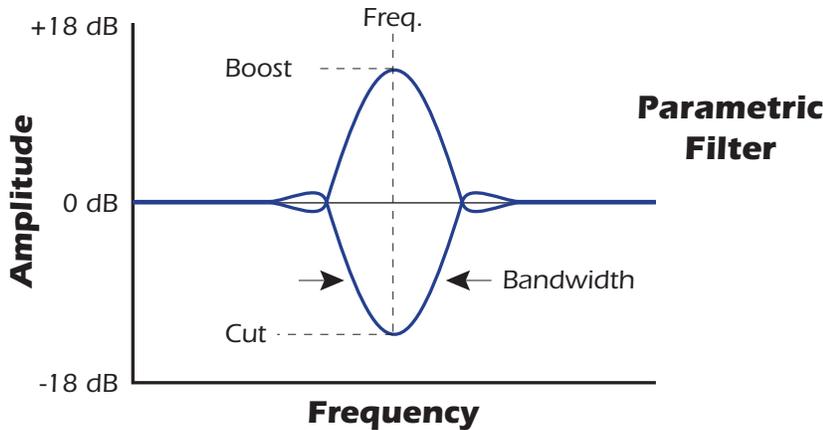
Using a filter, we now have a way to control the harmonic content of a sampled sound. As it turns out, even a simple low pass filter can simulate the response of many natural sounds.

For example, when a piano string is struck by its hammer, there are initially a lot of high frequencies present. If the same note is played softer, there will be fewer of the high frequencies generated by the string. We can simulate this effect by routing keyboard velocity to control the low pass filter. The result is expressive, natural control over the sound.

If an envelope generator is used to control the cutoff frequency of a filter, the frequency content can be varied dynamically over the course of the note. This can add animation to the sound as well as simulate the response of many natural instruments.

Parametric Filters

A more complex type of filter is called a parametric filter or Swept EQ. A parametric filter allows control over three basic parameters of the filter. The three parameters are: Frequency, Bandwidth, and Gain. The Frequency parameter allows you to select a range of frequencies to be boosted or cut, the Bandwidth parameter allows you to select the width of the range, and the Gain parameter either boosts or cuts the frequencies within the selected band by a specified amount. Frequencies not included in the selected band are left unaltered. This is different from a band pass filter which attenuates (reduces) frequencies outside the selected band.

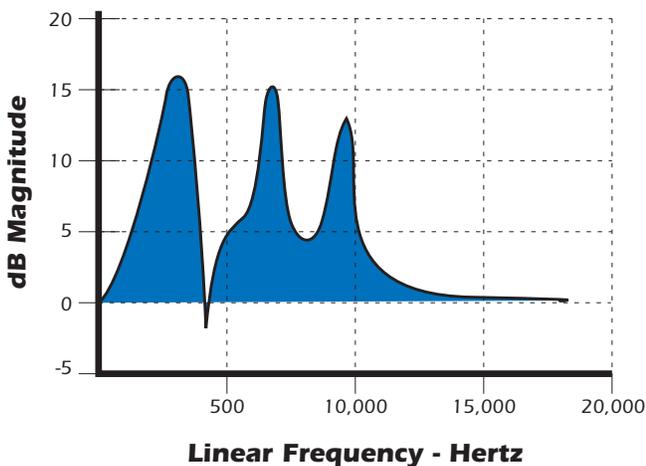


Another parameter sometimes used on a parametric filter is Shelving. Shelving simply widens the passband so that it extends to the limit of the upper or lower frequency range.

The parametric filter is quite flexible. Any range of frequencies can be either amplified or attenuated. Several parametric sections are often cascaded in order to create complex filter response curves.

If four parametric filter sections were cascaded, it would be possible to create the following complex filter response.

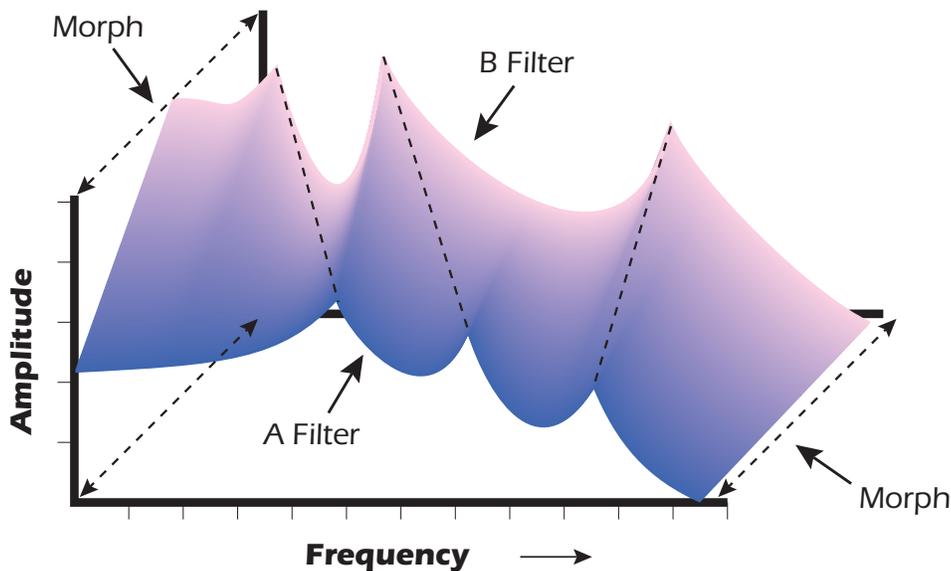
4 Parametric Equalizers



Many natural instruments have complex resonances which are based on their sound-board or tube size. The resonance shown above would be impossible to create using a normal synthesizer filter.

The Z-Plane Filter

A Z-plane filter is a filter which can change its function over time. In a Z-plane filter, we start with two complex filter types and interpolate between them using a single parameter. Refer to the diagram below.



The Z-plane filter has the unique ability to change its function over time.

Filters A and B represent two different complex filters or “frames”. By changing a single parameter, the Morph, many complex filter parameters can now be changed simultaneously. Following along the Morph axis you can see that the filter response smoothly interpolates between the two filters. This is the essence of the Z-plane filter. Through the use of *interpolation*, many complex parameters are condensed down into one manageable entity.

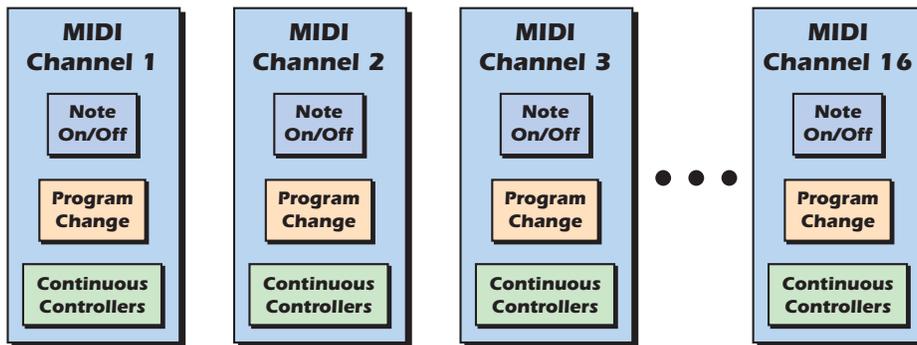
Consider, as an example, the human vocal tract, which is a type of complex filter or resonator. There are dozens of different muscles controlling the shape of the vocal tract. When speaking, however, we don't think of the muscles, we just remember how it feels to form the vowels. A vowel is really a configuration of many muscles, but we consider it a single object. In changing from one vowel to another, we don't need to consider the frequencies of the resonant peaks! You remember the shape of your mouth for each sound and interpolate between them. This is similar to what the Z-plane filter does.

This Z-plane filter sweep can be controlled by an envelope generator, an LFO, modulation wheels or pedals, keyboard velocity, key pressure, etc. In fact, any of the modulation sources can control the Z-plane filter.

MIDI Channels & Real-time Controls

The MIDI real-time controllers may seem confusing at first, but they are really very simple once you understand them. You probably already know that there are 16 MIDI channels per MIDI cable. Each MIDI channel uses three basic types of messages; note on/off, preset changes, and continuous controller messages. Your MIDI keyboard, in addition to telling Emulator X which note was played, can also send aftertouch real-time control information, which simply means control occurring in real-time or “live.” (You may be using a MIDI device other than a keyboard, but for simplicity's sake we'll presume that you are using a keyboard.) Real-time control sources include such things as controller knobs, pitch wheels or touchstrips, modulation wheels, control pedals and aftertouch. These are used to add more expression or control.

The front panel control knobs send out real-time controller information on separate continuous controller numbers. There is a set of 95 continuous controller numbers for each MIDI channel. Some continuous controllers, such as modulation wheel, volume, and pan have standardized numbers. For example, volume is usually sent on continuous controller #7. The front panel control knobs are programmed to specific MIDI controls.

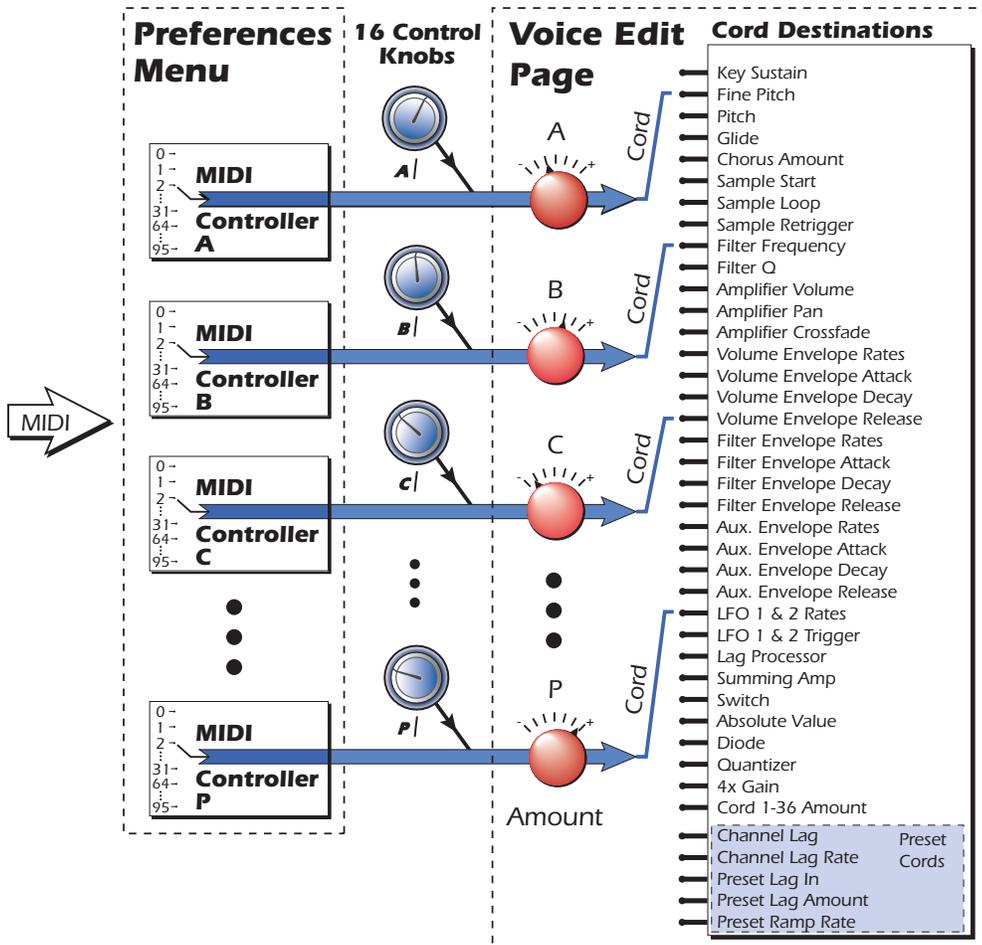


Any MIDI controller can be routed to any modulation destination. First, you have to know which controller numbers your keyboard transmits. Most modern MIDI keyboards let you select a controller number for each control on the keyboard. For example, it may let you select a number from 0-31 for the data slider. The realtime controller numbers that the keyboard transmits must match the numbers Emulator X is receiving, otherwise nothing will happen when you move the controls.

Suppose you wanted to send the four data sliders on your master keyboard. Emulator X can handle up to 16 MIDI controllers (A-P) of your choosing. “MIDI A-P” are simply names for the internal connections that link external MIDI continuous controllers to the Cords page. There are two parts to the connection. First, MIDI controller numbers are assigned to the letters A-P in the Controllers dialog box. Next, the letters A-P are connected to synthesizer control parameters in the PatchCord section of the Preset Edit window. The Cord Amount scales the amount of each controller by a positive or negative value.

Most factory presets have the MIDI A-P controls connected to standard synthesizer functions (labelled in the Multisetup page). By choosing any four of the 16 standard functions, the four sliders on your keyboard will work the same on every preset. The chart on the following page shows how this might work if your keyboard transmitted the slider settings on MIDI controllers 21-24.

Control	MIDI Controller #	Routing Letter	Standard Function
Slider 1	21	A	Controls Filter Frequency
Slider 2	22	B	Controls Filter Resonance
Slider 3	23	C	Controls Filter Attack
Slider 4	24	D	Controls Filter Decay



MIDI A-P are internal connections which carry MIDI continuous controller data. Assign a MIDI Continuous Controller numbers to a letter A-P in the Preferences menu, then assign the same letter to a modulation destination in the Voice Processing window.

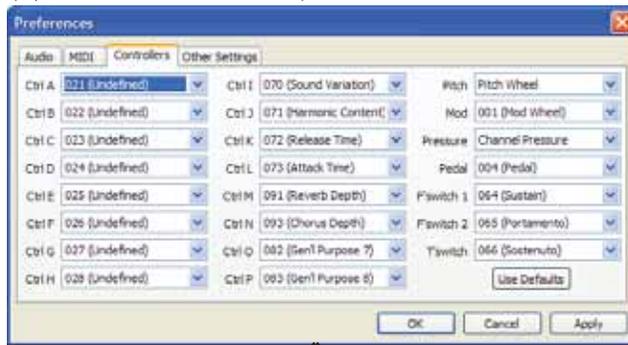
Assigning MIDI Controller Knobs

MIDI Device
 (Keyboard)



Controller
 Knobs

Preferences (Options, Preferences Controllers)



Select the CC numbers
 you wish to receive.

Continuous controller numbers
 are assigned Letters (A-P)
 & Labels (mod wheel, etc.)

Voice Processing Cords

Preset Global Cords



Assign the controller letters
 to Preset destinations or..

or



Assign the controller letters
 to Synth destinations.

MIDI Program Change Commands

Emulator X receives MIDI Program Change commands on each of its 64 MIDI channels. MIDI program changes select Emulator X *presets*. The “Receive Program Changes” button must be turned on (depressed) in the Preferences dialog box in order to receive program changes or bank select commands.

MIDI Bank Select Commands

Emulator X accepts MIDI bank select commands using MIDI continuous controller numbers. Continuous Controller (CC) 0 is the MSB (most significant byte) and CC 64 is the LSB (least significant byte). Normally you send both the MSB and LSB controllers to implement a bank change.

Emulator X remembers the MSB and the LSB that were last sent (or last changed from the front panel). For example, if you have already set the Bank MSB to 04, you need only send the LSB to change banks within the bank.

The selected bank remains selected until you change it (either via MIDI or by changing the bank from the front panel). Standard MIDI Program Change commands select from 128 presets within the selected bank.

Background: Bank Select Commands

When the original MIDI specification was developed, no commercially available synthesizer had more than about 100 preset locations. At that time being able to select up to 128 presets didn't seem like much of a limitation. So it was that the original MIDI specification provided for the selection of up to 128 presets.

As synthesizers came to have even *more* presets the MIDI specification was later amended to include *Bank Select Commands*. It was decided that Bank Select Commands would use Continuous Controllers 0 and 32 to allow the selection of up to 16,384 banks of 128 presets (over two million presets).

Because Bank Selects are implemented using Continuous Controllers, the Bank Selections can be made *per channel*. (This is getting better and better.) For each MIDI channel, you can select any of 16,384 banks and then one of the 128 presets in the bank. Of course no synthesizer has 16,384 banks (yet), but hey, it's nice to know it's possible (for that really BIG project).

MIDI Modes

There are three MIDI modes used by Emulator X. Multi Mode is the default mode. Select MIDI Modes from the toolbar buttons or from the Taskbar, Options menu.

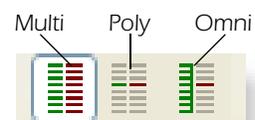
- **Omni Mode:** Voice messages are received on any and all MIDI channels and played on the basic channel. This mode is the lowest common denominator of MIDI. It is designed so that you always hear something. Also called “Mode 1”.
- **Poly Mode:** Voice messages are received only on one MIDI channel (the basic MIDI channel). Also called “Mode 3”.
- **Multi Mode:** Voice messages on each MIDI channel play on the preset assigned to that channel. Emulator X has the ability to play a different preset on each of 64 MIDI channels (using two MIDI ports of 16 channels each).

The **Basic MIDI Channel** can be identified by the red channel number in the Multi page. Click on any channel number to make it the basic channel.

■ **Tip:** Select “Show Bank and Program Numbers” in the toolbar to view program and bank numbers in the preset list.

Bank and Program numbers are displayed in Single Preset mode screen.

◆ The general term for “Preset” in MIDI-speak is “Program”. Same thing.



13 - Appendix

Translator Emulator X Converter

The Translator Emulator X Converter by Chicken Systems, Inc. is a special application that enables you to convert most popular sampler formats into the Emulator X's .exb file format, for use in Emulator X. Any source media may be used, such as CD-ROM's, fixed hard drives, ZipDisks, all removable media, magneto-optical drives, and all others. Exception: floppy media is not supported, however, see notes on possible workarounds.

Supported File Types

The Translator Emulator X Converter can convert the following types of files into Emulator X format:

- Akai S-1000/3000
- Akai MPC
- Emu EIII/ESI/Emax
- Tascam GigaStudio 1 + 2
- SoundFont
- Native Instruments Kontakt 1 + 2
- Steinberg HALion I and II

If you don't have sounds for any of these products, you won't need to use Translator. Keep the application installed in case you ever want to convert file formats.

► To Use Translator for Emulator X:

1. Install the Translator Emulator X application. This application is available from Chicken Systems, Inc. (www.chickensys.com) if it wasn't included in your E-MU software bundle.
2. Follow the instructions provided with the Translator application.

► **IMPORTANT: You don't have to convert Emulator IV banks, WAVE, AIFF Mp3 or REX files!**

1. Simply locate the desired EIV bank using the System Tab, then Open or Merge the bank.
2. Drag and drop REX 2 (.rx2, Recycle Export) .wav, .aiff, or mp3 files over the Emulator X icon to merge them into the bank. You can also Import them using the File, Import Samples... menu item.

► **Note about EOS Banks Saved on DOS Floppies**

Some EOS banks saved onto DOS floppies may end up with their 16-bit byte-swapped, LSB for MSB. The samples in these banks, when loaded into Emulator X, will typically sound like white noise. Emulator X provides a method by which to repair the data as it is loaded. To do this, simply rename the bank file so that the extension reads, ".b4e".

Emulator X VSTi

Launching the VSTi Application

The following instructions apply to Cubase LE 4. Other host applications will have their own unique methods for launching a VSTi application. If you are using a host application other than Cubase LE, consult the documentation that came with your host application. Also see the Emulator X Advanced Applications Guide at: www.emu.com.

1 - Setting up Cubase LE4

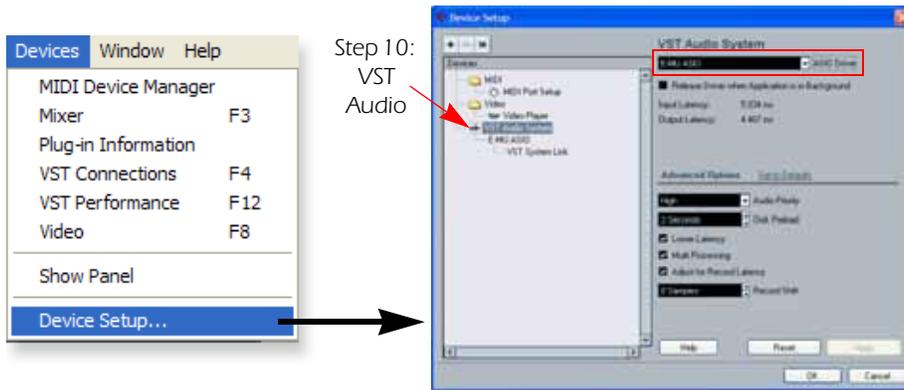
Follow these instructions carefully to ensure that Cubase LE 4 runs smoothly the first time. Cubase LE 4 will remember these settings, so you'll only have to do this once.

1. Open Cubase LE 4 from the Start menu. An ASIO multimedia driver test dialog box will pop up to ask if you want to run the ASIO test. Choose **No**, because you won't be using the driver anyway.
2. Select **New Project** from the **File** menu.
3. Select **Empty** and click **OK**.
4. A **Select Directory** pop-up dialog box will appear. Choose a location on your hard disk where you want to store your audio files, then click **OK**.
5. The Cubase LE 4 Project window appears.
6. Select **Device Setup...** from the **Devices** menu.



■ **Tip:** When running multiple VSTis, distribute the CPU % among the VSTis. Example: If one VSTi does not glitch at 60%, set the CPU% of two VSTis so they add up to 60%.

■ If you have two or more hard disks, it's better to store audio files on a disk that isn't running your OS.



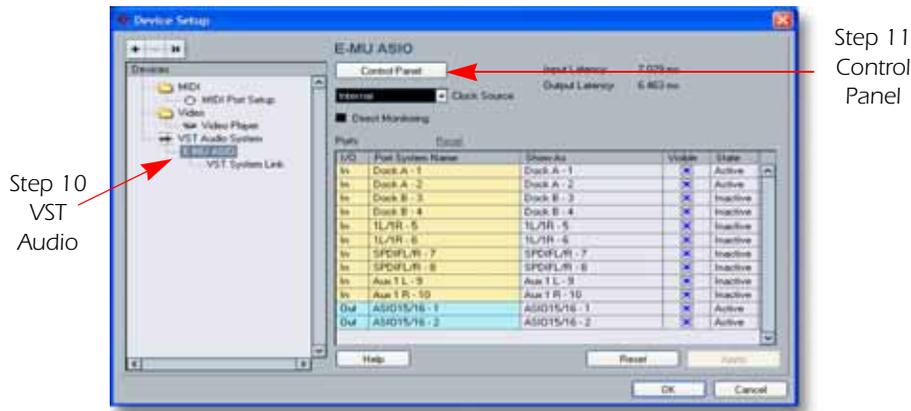
Step 10:
VST
Audio

7. Select **VST Audio System** from the left pane.
8. Select the ASIO Driver. Choose your E-MU interface from the list. Choose: **E-MU ASIO** (0404, 1212, 1820, 1616 Digital Audio Systems), **ASIO E-MU 0404|USB**, **ASIO E-MU 0202|USB**, or **ASIO E-MU TRACKER Pre|USB**.
9. A pop up dialog box asks you if you want to keep or switch the ASIO driver. Select **Switch**.

▼ **DO NOT** select the "ASIO Multimedia Driver" or the "ASIO Direct X Full Duplex Driver."

Buffer Latency Setting

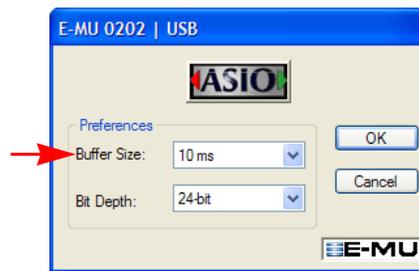
- Click on the name of your E-MU interface that appears immediately below **VST Audio System** in the Devices pane. Refer to the following screen.



Step 10
VST
Audio

Step 11
Control
Panel

- Click the **Control Panel** button. The pop-up dialog box shown at right appears.
- Set the **ASIO Buffer Latency** as low as your computer will allow and click **OK**. (10ms is a good starting point.) A low latency setting is important to assure fast response when using virtual instruments and to minimize delay when monitoring through Cubase. If you hear crackles or other audio problems, try increasing the Buffer Size.



◆ E-MU 0202 Control Panel is shown at left. The Bit Depth field does not appear with 0404PCI, 1212, 1820 or 1616.

- Close the Device Setup screen by clicking **OK**.

Setting up a New Project

- Select **New Project** from File menu in Cubase LE 4. A pop-up Template selection dialog box appears. Select **Empty**, then click **OK**.
- Another pop-up dialog box appears asking you to select the directory in which your audio files will be stored. Choose a location to store your audio files and click **OK**.

- Select **Project Setup** (Shift +S) from the **Project** menu. This is where you set the Sample Rate and Record Format (bit depth), among other things. Set the Record Format to **24 Bit** and the Sample Rate to **44.100 kHz**.

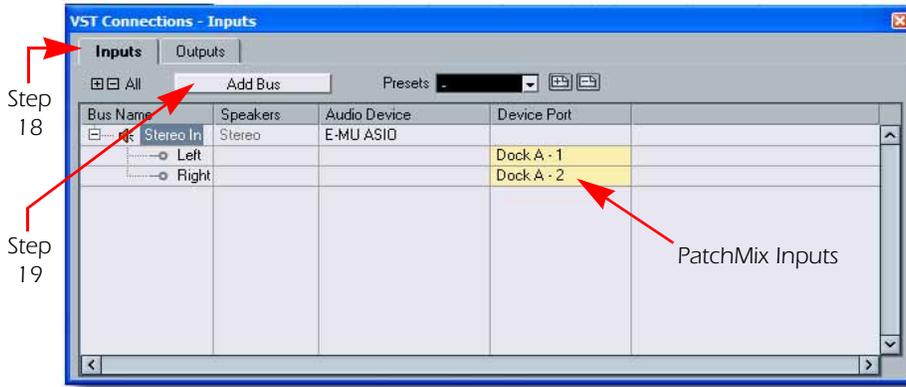
Optional - Add Additional Input Busses

If you are using an audio interface that has more than two inputs (0404 USB, 0404, 1212, 1616, 1820), the following instructions will allow you to select these extra inputs to feed your audio tracks.

- Select **VST Connections** (F4) from the **Devices** menu.



◆ If the Cubase LE 4 application crashes for any reason, it is recommended that you re-boot the computer.



18. Select the **Inputs** Tab, revealing the window shown above.

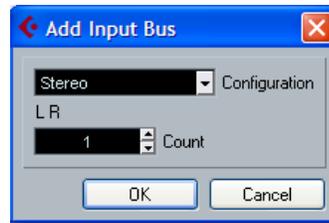
19. Click the **Add Bus** button. The pop-up dialog box shown at right appears.

20. Select the number of mono or stereo input busses you want, then click **OK**.

21. Additional mono or stereo busses appear in the list. Note that the actual PatchMix input label appears in the Device Port field.

22. Close the window by clicking the close

box. The inputs you added will now be available for selection in the Track Input field of the Inspector pane.



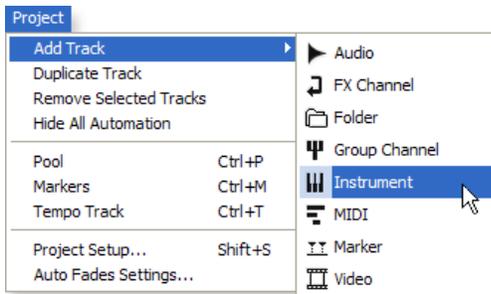
■ You can also Rename your Input Busses here to avoid confusion when assigning Inputs to tracks.

2- Recording a MIDI Track using Emulator X3

You'll need a MIDI interface and a MIDI keyboard (or other MIDI input device) for this tutorial.

Make the Connections

1. Connect the MIDI out of your MIDI keyboard to the MIDI input of your MIDI interface.
2. From the **Project** menu, select **Add Track, Instrument**.



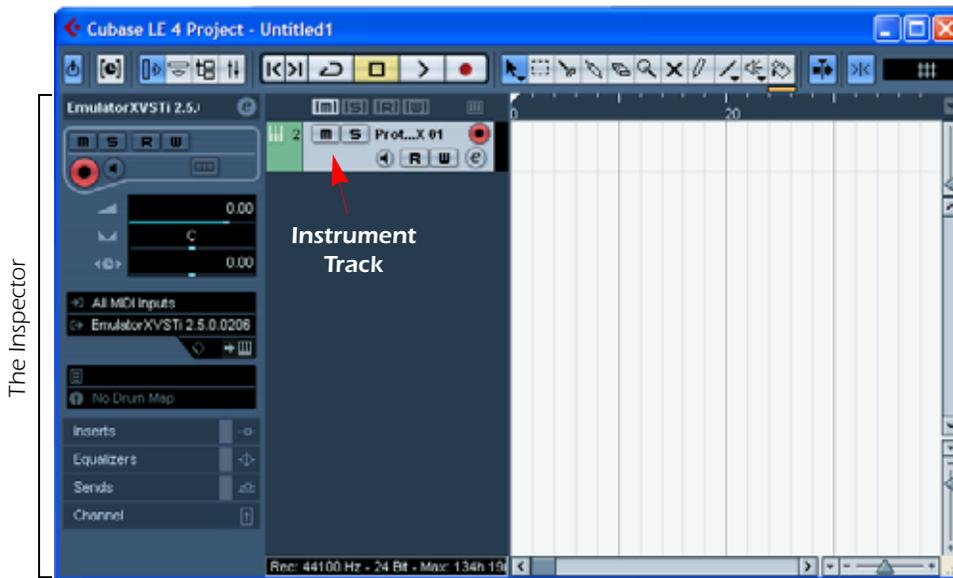
◆ Emulator X3 can also run as a standalone application.

3. A pop-up dialog box appears asking you to select a VST Instrument. Select Emulator X3. Select a Count of 1. Click OK.



◆ Selecting a Count of 2 or more, instantiates multiple copies of Emulator X3, which results in very inefficient usage of your CPU. (See the note on [page 325.](#))

After a short delay, the Cubase Project Window should now look more or less like the one below with one Instrument track:



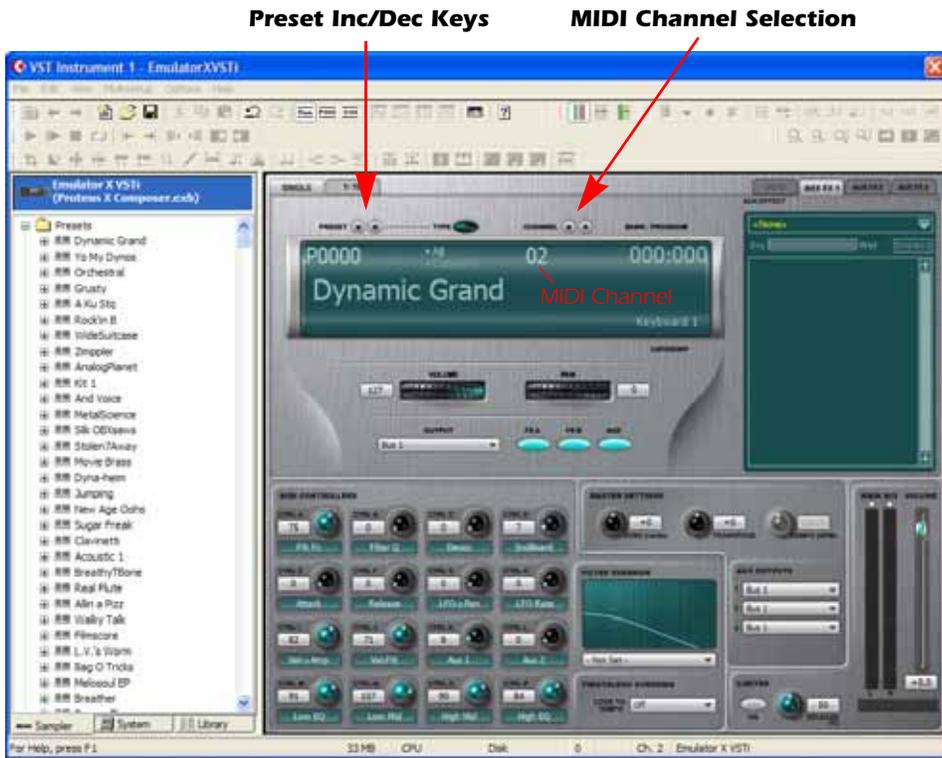
Open VX & Load a Bank

4. Since Emulator X3 uses samples as the basis of its sound, you need to load a bank of samples before it can be played. The Edit Instrument button, located in the Inspector section of the window, allows you to edit the virtual instrument.



Edit Instrument Button

- Click the **Edit Instrument** button. The Emulator X3 main window appears.



- Now we can load the Proteus X Composer bank. Select **Proteus X Composer** from the File menu. The bank is installed here: "Program Files\Creative Professional\E-MU Sound Central\Proteus X Composer." Loading takes a few seconds.
- Change the Preset using the Inc/Dec keys.** You also have to select a preset before you can hear anything. There are 1024 different presets (sounds) in this huge bank.
- Bring up the **mini keyboard**  by clicking the icon on Emulator X3 and play a few notes. You should be hearing sound.
 - If the sound volume is very low, you can decrease the Headroom of Emulator X3. (Options, Preferences, Headroom/Boost) Keep in mind that with less headroom, the Emulator X3 will be more prone to clipping when multiple channels are played.
- Play your MIDI controller** and verify that it plays Emulator X3. If not, check to make sure your MIDI keyboard is set to the same MIDI channel number as Emulator X3 (*probably channel 1*).
- Try out the MIDI Controller knobs on your MIDI keyboard. In order for these to work, the continuous controller numbers of your keyboard knobs must match those on Emulator X3. (Options, Preferences, Controllers tab).
- Feel free to play around for awhile and don't worry about losing anything. Nothing is made permanent until you **Save** the bank, so have fun.

■ Increase the headroom setting (more positive) to increase the volume and decrease the amount of available headroom.

To Record a MIDI Track

12. Make sure the **Record Enable** button on the MIDI track is on (it should be on by default).



13. Click **Record** on the Cubase LE Transport control and start playing your MIDI controller.



14. Press **Stop** when you're finished recording the first track.
15. Press the **Go To Start** button. ⏪
16. Press **Play** > on the Cubase Transport to play back your track.

To Record a MIDI Track on another MIDI Channel

NOTE: Emulator X3 VSTi can play back up to 16 MIDI tracks at once, however, the LE version of Cubase 4 only supports a single MIDI channel per VSTi. You *could* load another instance of the Emulator X3 for each channel, but this is a very inefficient usage of your CPU resources and is NOT recommended. The full version of Cubase 4 does support multiple MIDI channels on a single VSTi.

Program Changes

Cubase LE 4 allows you to insert **MIDI Program Change** messages in the MIDI sequence. (Key Editor, List Editor, Insert Program Change) This trick makes it possible to use more than one Emulator X3 preset in a song.

On Your Own

Cubase LE 4 and Emulator X3 include excellent documentation and help files. Don't overlook the Emulator X Advanced Application Guide, (located at www.emu.com) which contains 43 in-depth tutorials on all aspects of the Emulator X. Take the time to learn all the features of these powerful programs so you can expand your musical boundaries.

Multisetup is Saved in the Recording Application

The current Multisetup is automatically saved when you save your song in your recording application such as Cubase or Sonar.

Simply set up the Multisetup the way you want it, or select one from the Tree, then save your song. The Multisetup will be saved along with your song.

Checklist for Launching Emulator X VSTi

Cubase LE

Choose the following settings in the Options menu.

- **Options, Audio Setup, System**

Audio I/OE-MU ASIO or E-MU USB

- **Options, MIDI Setup, System**

MIDI InputSelect the MIDI port your KYBD is using

If you're using PatchMix DSP..

Make sure you have an ASIO 1 & 2 input strip in the mixer.

Keyboard Shortcuts

File Menu

New	Ctrl + N
Open	Ctrl + O
Merge	Ctrl + M
Save	Ctrl + S
Acquire Samples	Ctrl + Q

Edit Menu

Undo.....	Ctrl + Z
Redo	Ctrl + Y
Cut.....	Ctrl + X
Cut.....	Ctrl + X
Copy	Ctrl + C
Paste	Ctrl + V
Duplicate.....	Ctrl + U
Delete	Del
Select All.....	Ctrl + A

Sample Edit Menu

Select Loop.....	Ctrl + L
Select All.....	Ctrl + A
External Edit.....	Ctrl + E
Beat Analysis	Ctrl + B
Shift Beats Right	Shift + →
Shift Beats Left	Shift + ←

Transport Menu

Play from Cursor	Space
Play from Start	Ctrl + Space
Go To Start	Home
Go To End	End
Go To Loop Start.....	Ctrl + Home
Go To Loop End	Ctrl + End
Go To Beginning of Selection.....	[
Go To End of Selection.....]

Tree

Expand Folder or Node	Ctrl + →
Collapse Folder or Node	Ctrl + ←
Move Up Through Items	↑
Move Down Through Items	↓

Zoom

Zoom In	Page Up
Zoom Out	Page Down
Zoom In Vertically.....	Ctrl + Page Up
Zoom Out Vertically.....	Ctrl + Page Down
Zoom to Selection.....	F10
Zoom to All.....	F11
Zoom to Sample	F12

Sample Loop Menu

Make Loop From Selection	Ctrl + Shift + L
Duplicate Loop	Ctrl + Shift + D
Move Start To Previous Zero	Left Ctrl + ←
Move Start To Next Zero.....	Left Ctrl + →
Move End To Previous Zero.....	Right Ctrl + ←
Move End To Next Zero	Right Ctrl + →
Autocorrelate Loop	Ctrl + K

Options

All Sound Off.....	Ctrl + Pause
--------------------	--------------

Window

Close	Alt + F4
-------------	----------

Presets

New Preset	Ctrl + W
Open Voices	Ctrl + Enter
Open Sample.....	Alt + Enter
New Voice.....	Ctrl + W
New Sample Zone	Alt + W

Links

New Link.....	Ctrl + W
---------------	----------

View

Refresh	F5
Keyboard.....	F9

Knobs, Sliders, and Wheels

(Place the cursor over the knob, slider or wheel)

Center Control (Pan, etc.).....	Alt + Click (Number key 5 also centers control when Num Lock is off)
Minimum Setting	Home
Maximum Setting.....	End

Drag & Drop

You can use simple “Drag & Drop” operations for many Emulator X functions.

Drag & Drop Operations

From	Drag & Drop To...	Action
Bank File in Windows	Emulator X Icon	Load or Merge Bank
Bank File in Windows	Emu-X System Folder	Copy Move Bank
Audio File in Windows	Instrument	Import
Audio File in Windows	Samples Folder	Import
Audio File in Windows	Keyboard Key in Voices and Zones window	Import
Audio File in Windows	Keyboard Display in Emulator X	Import to New Voice Mapped to Key
Audio File in Windows	Samples folder in the Tree.	Import
Emulator X Bank File	Emulator X Icon	Load/Merge
Emulator X Bank File	Folder in Windows	Copy/Move
Emulator X Bank File	Emu-X System Folder	Copy/Move
Emulator X Bank Preset	Emulator X Icon	Merge
Emulator X Bank Preset	Presets Folder	Merge
Emulator X Bank Preset	Multisetup Slot	Merge & place
Emulator X Bank Preset	Emu-X System Folder	Export as Bank
Emulator X Bank Preset	Link	Merge & Replace Link
Emulator X Bank Preset	Link (null space)	Merge & Add Link
Emulator X Bank Sample	Samples Folder	Import
Emulator X Bank Sample	Preset Voice	Merge & add to Voice
Emulator X Bank Sample	Emu-X System Folder	Export as Bank/Wave/AIFF
Emulator X Bank Sample	Preset Zone	Replace Zone
Emulator X Bank Sample	Preset Voice (null space)	Add Voice
Emulator X Bank Multisetup	Emulator X Icon	Merge
Emulator X Bank Multisetup	Multisetup Folder	Merge
Preset	Multisetup Preset Slot	Place
Preset	Emu-X System Folder	Export as Bank
Preset	Link	Replace Link

Drag & Drop Operations

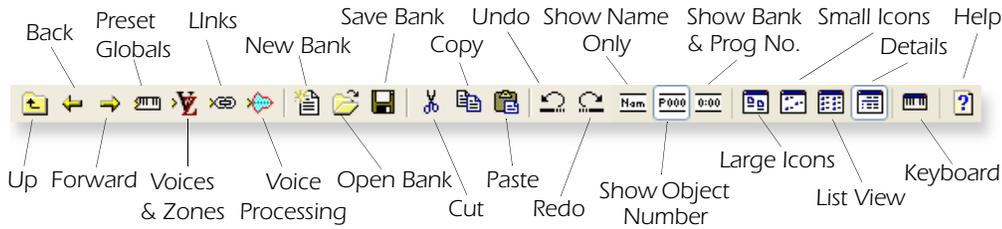
From	Drag & Drop To...	Action
Preset	Link (null space)	Add Link
Sample	Preset Voice	Add Zone to Voice
Sample	Preset Voice (null space)	Add Voice
Sample	Preset Zone	Replace Zone
Sample	Emu-X System Folder	Export as Bank/Wave/AIFF
Sample	Keyboard Key in Voices and Zones window	Add Voice
Multisetup	Instrument	Load
Instrument	Multisetup Folder	Save
Instrument	Multisetup	Replace
E4 Bank File	Instrument	Load/Merge
E4 Bank File	Emu-X System Folder	Copy/Move
E4 Bank Preset	Instrument	Merge
E4 Bank Preset	Presets Folder	Merge
E4 Bank Preset	Multisetup Preset Slot	Merge & Place
E4 Bank Preset	Folder in Windows	Load/Merge
E4 Bank Preset	Emu-X System Folder	Export as Bank
E4 Bank Preset	Link	Merge & Replace Link
E4 Bank Preset	Link (null space)	Merge & Add Link
E4 Bank Sample	Instrument	Merge
E4 Bank Sample	Samples Folder	Merge
E4 Bank Sample	Emu-X System Folder	Export as Bank/Wave/AIFF
E4 Bank Sample	Preset Zone	Replace Zone
E4 Bank Sample	Preset Voice (null space)	Add Voice
E4 Bank Multisetup	Instrument	Merge
E4 Bank Multisetup	Multisetup Folder	Merge
E4 Bank File (CD)	Instrument	Load/Move
E4 Bank File (CD)	Emu-X System Folder	Copy/Move

Drag & Drop Operations

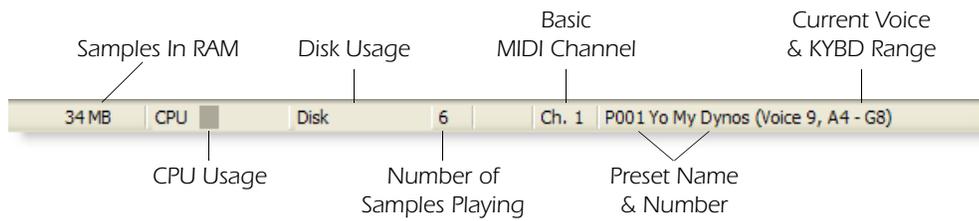
From	Drag & Drop To...	Action
E4 Bank Preset (CD)	Instrument	Merge
E4 Bank Preset (CD)	Presets Folder	Merge
E4 Bank Preset (CD)	Multisetup Preset Slot	Merge & Place
E4 Bank Preset (CD)	Emu-X System Folder	Export
E4 Bank Preset (CD)	Link	Merge & Replace Link
E4 Bank Preset (CD)	Link (null space)	Merge & Add Link
E4 Bank Sample (CD)	Instrument	Merge
E4 Bank Sample (CD)	Samples Folder	Merge
E4 Bank Sample (CD)	Preset Voice	Merge & Add to Voice
E4 Bank Sample (CD)	Emu-X System Folder	Export as Bank/Wave/AIFF
E4 Bank Multisetup	Instrument	Merge
E4 Bank Multisetup	Multisetup Folder	Merge

Toolbar Functions

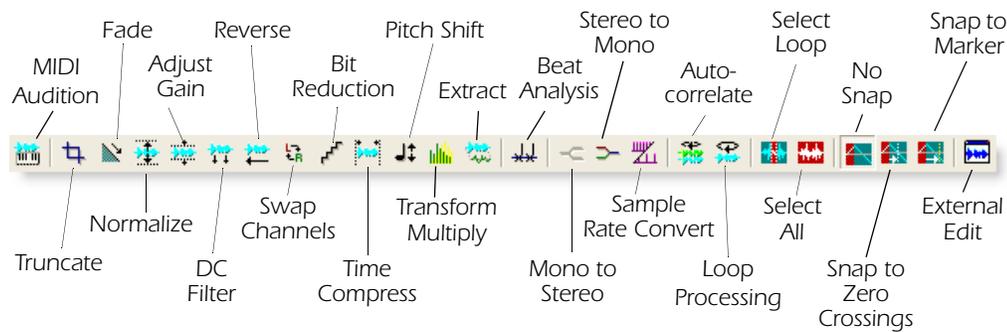
Standard



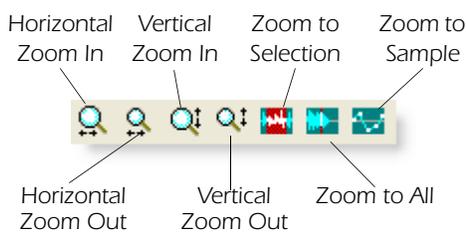
Status Bar



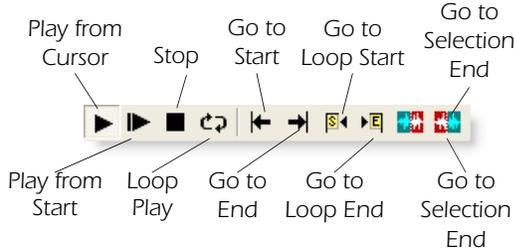
Sample Edit



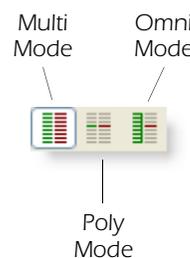
Sample Zoom



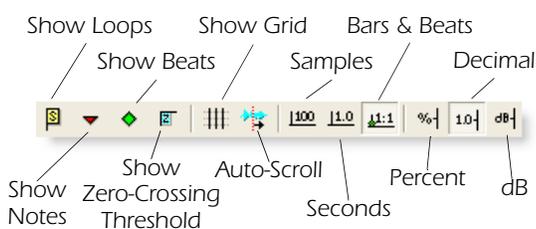
Sample Transport



MIDI Mode



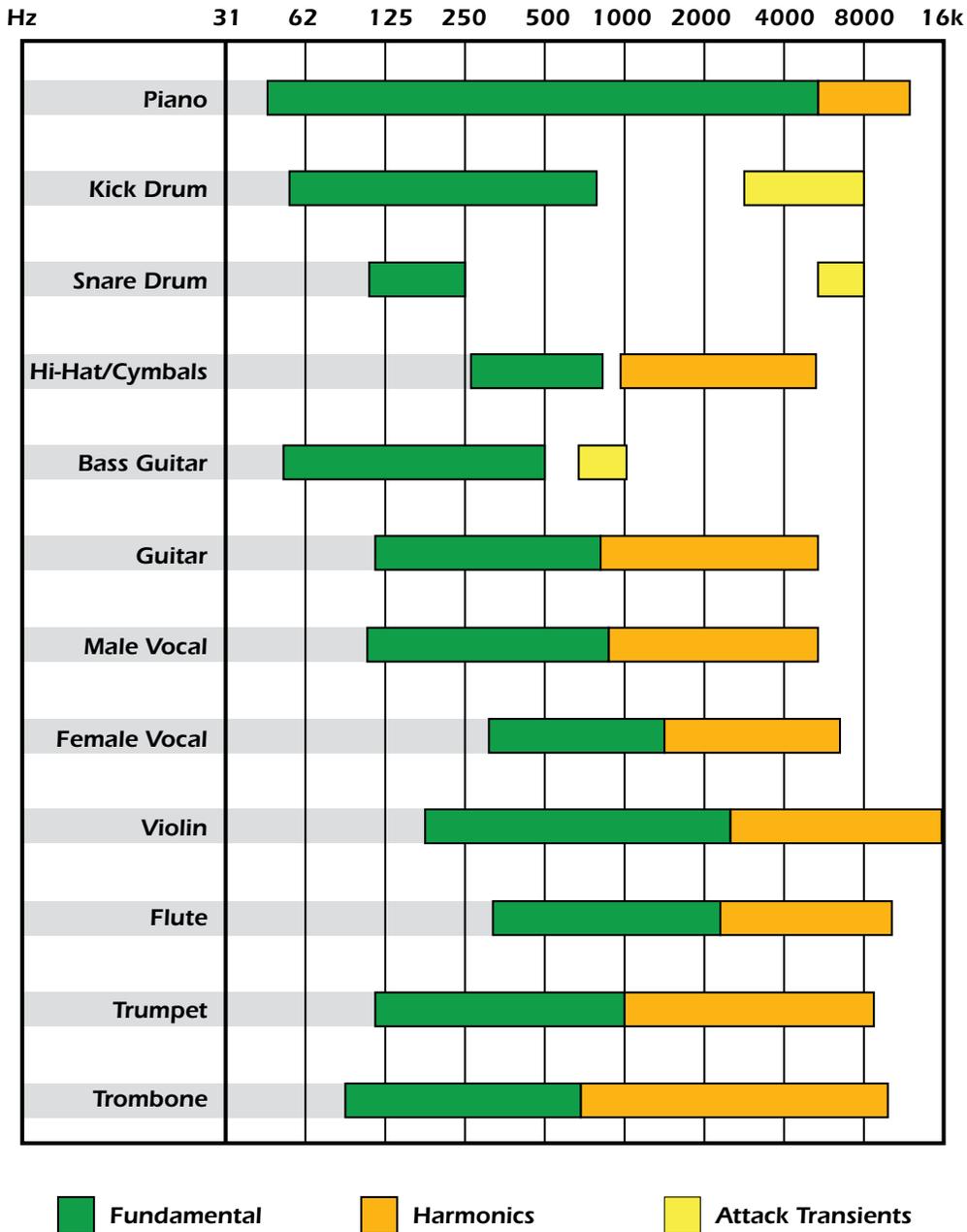
Sample View



MIDI Key Number to Note Chart

	0	1	2	3	4	5	6	7	8	9
1s		C-2	C#-2	D-2	D#-2	E-2	F-2	F#-2	G-2	G#-2
10s	A-2	A#-2	B-2	C-1	C#-1	D-1	D#-1	E-1	F-1	F#-1
20s	G-1	G#-1	A-1	A#-1	B-1	C-0	C#0	D0	D#0	E0
30s	F0	G0	G#0	A0	A#0	B0	C1	C#1	D1	D#1
40s	E1	F1	F#1	G1	G#1	A1	A#1	B1	C2	C#2
50s	D2	D#2	E2	F2	F#2	G2	G#2	A2	A#2	B2
60s	C3	C#3	D3	D#3	E3	F3	F#3	G3	G#3	A3
70s	A#3	B3	C4	C#4	D4	D#4	E4	F4	F#4	G4
80s	G#4	A4	A#4	B4	C5	C#5	D5	D#5	E5	F5
90s	F#5	G5	G#5	A5	A#5	B5	C6	C#6	D6	D#6
100s	E6	F6	F#6	G6	G#6	A6	A#6	B6	C7	C#7
110s	D7	D#7	E7	F7	F#7	G7	G#7	A7	A#7	B7
120s	C8	C#8	D8	D#8	E8	F8	F#8	G8		

Frequency Range of Common Instruments



▼ E4/E3 Library CDs use a proprietary format and can only be read using SCSI/ATAPI drives.

MIDI Implementation Chart

MIDI Information	Transmitted	Recognized	Remarks
MIDI Channels	No	1-16, 17-32, 33-48, 49-64	64 MIDI Channels on 4 MIDI ports
Note Numbers	No	0-127	
Program Change	No	0-127	
Bank Select Response?	No	Yes	MSB + LSB
Modes: Omni (Mode 1)	No	No	Responds to all MIDI channels
Mono (Mode 2)	No	No	
Poly (Mode 3)	No	No	
Mode 4 (Y/N)	No	No	
Multi (Mode 5)	No	No	
Note On Velocity	No	Yes	
Note Off Velocity	No	Yes	
Channel Aftertouch	No	Yes	
Poly (Key) Aftertouch	No	No	
Pitch Bend	No	Yes	
Active Sensing	No	No	
System Reset	No	No	
Tune Request	No	No	
System Exclusive	No	No	
Sample Dump Standard	No	No	
File Dump	No	No	
MIDI Tuning	No	No	
Master Volume	No	Yes	
Master Balance	No	No	
Notation Information	No	No	
Turn GM1 System On	No	No	
Turn GM2 System On	No	No	
Turn GM1 System Off	No	No	
Other (See Remarks)	No	No	
NRPNS	No	No	
RPN 00 (Pitch Bend Sensi.)	No	No	
RPN 01 (Chan. Fine Tune)	No	No	
RPN 02 (Chan Coar. Tune)	No	No	
RPN 03 (Tuning Prog Sel.)	No	No	
RPN 04 (Tuning Bank Sel.)	No	No	
RPN 05 (Mod Depth Rang)	No	No	
MIDI Timing & Sync	No	No	
MIDI Clock	Yes	Yes	
Song Position Pointer	No	No	
Song Select	No	No	
Start	No	No	
Continue	No	No	
Stop	No	No	
MIDI Time Code	No	No	
MIDI Machine Control	No	No	
MIDI Show Control	No	No	
General MIDI Compat?	No	No	
Is GM default mode?	No	No	
DLS compatible?	No	No	
Import DLS Files?	No	No	
Export DLS Files?	No	No	
Import Std MIDI files	No	No	
Export Std MIDI files	No	No	

MIDI Implementation Chart (part 2 - Controllers)

Control #	Function	Transmitted	Recognized	Remarks
0	Bank Select MSB	No	Yes	See Note
1	Mod Wheel MSB	No	Yes	*
2	Breath Cntl MSB	No	No	*
3		No		*
4	Foot Cntl MSB	No	No	*
5	Portamento MSB	No	No	*
6	Data Entry MSB	No	No	*
7	Chan Volume MSB	No	Yes	*
8	Balance MSB	No	No	*
9				*
10	Pan MSB	No	Yes	*
11	Expression MSB	No	Yes	*
12	Effect Cntl 1 MSB	No	No	*
13	Effect Cntl 2 MSB	No	No	*
14				*
15				*
16	GenPur Ctrl 1 MSB			*
17	GenPur Ctrl 2 MSB			*
18	GenPur Ctrl 3 MSB			*
19	GenPur Ctrl 4 MSB			*
20				*
21				*
22				*
23				*
24				*
25				*
26				*
27				*
28				*
29				*
30				*
31				*
32	Bank Select LSB			*
33	Mod Wheel LSB			*
34	Breath Cntl LSB			*
35				*
36	Foot Cntrlr LSB			*
37	Portamento LSB			*

MIDI Implementation Chart (part 2 - Controllers)

Control #	Function	Transmitted	Recognized	Remarks
38	Data Entry LSB			*
39	Chan Volume LSB			*
40	Balance LSB			*
41				*
42	Pan LSB			*
43	Expression LSB			*
44	Effect Cntl 1 LSB			*
45	Effect Cntl 2 LSB			*
46				*
47				*
48	Gen Pur Ctrl 1 LSB			*
49	Gen Pur Ctrl 2 LSB			*
50	Gen Pur Ctrl 3 LSB			*
51	Gen Pur Ctrl 4 LSB			*
52				*
53				*
54				*
55				*
56				*
57				*
58				*
59				*
60				*
61				*
62				*
63				*
64	Sustain Pedal	No	Yes	*
65	Portamento on/off	No	No	*
66	Sostenuto	No	No	*
67	Soft Pedal			*
68	Legato Footswitch			*
69	Hold 2			*
70	Variation			*
71	Timbre/Har Inten	Yes	Yes	*
72	Release Time	Yes		*
73	Attack Time	Yes		*
74	Brightness	Yes		*
75	Sound Control 6	Yes	Yes	*
76	Sound Control 7			*

MIDI Implementation Chart (part 2 - Controllers)

Control #	Function	Transmitted	Recognized	Remarks
77	Sound Control 8	Yes	Yes	*
78	Sound Control 9	Yes	Yes	*
79	Sound Control 10	Yes	Yes	*
80	Gen Purp Control 5	Yes	Yes	*
81	Gen Purp Control 6			*
82	Gen Pur Control 7	Yes	Yes	*
83	Gen Pur Control 8	Yes	Yes	*
84	Portamento Cntrl			*
85				*
86				*
87				*
88				*
89				*
90				*
91	Effects 1 Depth	Yes		*
92	Effects 2 Depth			*
93	Effects 3 Depth	Yes		*
94	Effects 4 Depth			*
95	Effects 5 Depth			*
96	Data Increment			
97	Data Decrement			
98	NRPN (LSB)			
99	NRPN (MSB)			
100	RPN (LSB)			
101	RPN (MSB)			
102				
103				
104				
105				
106				
107				
108				
109				
110				
111				
112				
113				
114				
115				

MIDI Implementation Chart (part 2 - Controllers)

Control #	Function	Transmitted	Recognized	Remarks
116				
117				
118				
119				
120	All Sound Off	No	Yes	See note ✱
121	Reset All Controllers	No	Yes	
122	Local Cntrl on/off	No	No	
123	All Notes Off	No	Yes	
124	Omni Mode Off	No	No	
125	Omni Mode On	No	No	
126	Poly Mode Off	No	No	
127	Poly Mode On	No	No	Poly mode is always on.
<p>NOTES: * Emulator X can receive ANY continuous controller number from 1 to 95. Because of Emulator X's powerful synth engine, many of the standard MIDI controllers can be user programmed to provide the desired function. A "Yes" response in this chart means that a controller is programmed by default in Emulator X.</p> <p>✱ Terminates all sound from the synthesizer and clears the effect processors of all sound data. The All Sound Off command is automatically initiated when the sequencer transport Stop is pressed. It can also be manually invoked from the Options menu or by pressing Ctrl+Pause on the computer keyboard.</p> <p>Other: Pan: -64 = hard left, +63 = hard right</p>				

Received Channel Commands

Channels number (n) = 0-15. Message bytes are represented in hex. All other numbers are decimal. Running Status is supported.

Command	Message	Comments
Note Off	8n kk w	
Note On	9n kk w	velocity 0 = note off
Key Aftertouch	An kk w	kk = 0-127 w = 0-127
Program Change	Cn w	0-127
Channel Aftertouch	Dn w	0-127
Pitch Bend	En ll mm	l = lsb, m = msb
Real-time Controller	Bn cc w	cc = 00-31, 64-95
Footswitch	Bn cc w	cc = 64-79, w \neq 64 = on
Volume	Bn 07 w	0-127
Pan	Bn 0A w	0=left, 127=right, 64=center
All Sound Off	Bn 78 00	turns all sound off
Reset All Controllers	Bn 79 00	ignored in omni mode
All Notes Off	Bn 7B 00	ignored in omni mode
Omni Mode Off*	Bn 7C 00	forces all notes & controls off
Omni Mode On*	Bn 7D 00	forces all notes & controls off
Mono Mode On (Poly Off)*	Bn 7E 00	forces all notes & controls off
Poly Mode On (Mono Off)*	Bn 7F 00	forces all notes & controls off
Bank Select MSB	Bn 00 bb	bb = bank MSB Go There!
Bank Select LSB	Bn 20 bb	bb = bank LSB Go There!

Special Notes:

- **From Omni Mode** Omni Off turns Poly On.
- **From Poly Mode** Omni On turns Omni On; Mono On turns Mono On.
- **From Mono Mode** Mono Off turns Poly On; Omni On turns Omni On.
- **From Multi Mode** Omni On turns Omni On; Omni Off or Mono Off turns Poly On; Mono On turns Mono On.
All other changes have no effect

Optimizing Your Computer for Emulator X

The following suggestions can help you configure your Windows machine for optimal performance. Since systems vary, we recommend that you note the original settings of any system parameters you alter so that you can restore them in the event that unexpected problems occur as you seek to optimize your system for Emulator X:

- Seriously consider dedicating a computer as your audio workstation. This computer can be optimized for best audio performance.
- If possible, avoid assigning slower devices to the IDE or serial ATA controller used by your audio drive.
- Get the fastest hard disk drive you can afford, as this is the single most important component affecting Emulator X performance. Many drives now offer 8 or 16 MB data buffers, which can also improve performance.
- Defragment your hard drive often.
- Perform a “clean” install of your operating system with only the minimum components and programs installed.
- Use System Restore or other restoration program such as “Ghost®” to keep your system uncluttered and fast.

SCSI and IDE Drive Tips and Considerations

When considering the purchase of a new hard disk drive for use with Emulator X, make sure that the new drive supports a sustained data transfer rate of at least 2 megabytes per second. While it is impossible to reliably calculate the needs of your computer's operating system — which must be considered in addition to Emulator X's requirements — you can get an approximate idea of the necessary speed of your hard disk drive based on the bit depth and number of voices you need.

▼ E4/E3 CDs use a proprietary format and can only be read using SCSI/ATAPI drives.

Data Transfer Rate Chart

16 bits	44.1 k sample rate	86.2 k per Channel per second
	48 k sample rate	93.8 k per Channel per second
24 bits	44.1 k sample rate	138 k per Channel per second
	48 k sample rate	150 k per Channel per second

- Regularly check your hard drive for fragmentation and de-fragment the drive when necessary. Your drive will operate most efficiently when it's not fragmented.
- If possible, use a two-drive system: one for audio, one for your system.

Index

Symbols

~, +, - modulation source polarity 160

Numerics

1-Band Para EQ 254
22-Tone Tuning 83
24-Tone Tuning 83
2nd Stage Gain 267
36-Tone Tuning 83
4-Band EQ 255
4x Gain 307

A

Absolute Value 307
Acquire Samples 281
Activate, software 19
Add 97
 a sample to a voice 60, 97
 a voice and sample 98
 preset 327
Add Input Busses, to Cubase LE 321
Add Track, Cubase LE 322
Adjust Gain, sample 187
ADSR Envelopes 133
Advanced Applications Guide 65, 299
Advanced Preference Settings 25
Aftersample, Mono 300
AIFF, import 171
Akai, converting files to Emulator X 319
al-Farabi 12, tuning 83
All Sound Off 278, 327
 MIDI command 338
Alternate Tunings 83
Amount, patchcord 314
Amp Envelope Depth 154
Amplifier
 controls 154
 envelope tutorial 63
 pan 154
 volume 154
Analog Oscillator Sound 130
Analyzing Beats 209

Architecture, description 35
Arpeggiator, using function generator as 140
ASIO
 audio drivers 23
 inputs for sampling 282
 outputs 236
ASIO Buffer Latency 321
Assign Group, voice 164
Audio Setup 23
Audition
 audio file from the import browser 171
 MIDI 185
 preset or sample from library 67
 presets 45, 47
 sample from disk 65
 under Vista 171
Auto Balance, sample placement 295
Autocorrelate Loop 178
Auto-Loop 296
Auto-Normalize 296
Aux FX 231
 adding 232
 bus routing diagram 75
 enable/disable 233
 select 240
 send amounts 240
 view buttons 73
Aux Output 75
 enable/disable 155, 237
 on/off 237
 select 237
Aux Sends 1-3 155
 on/off 73

B

Backward
 playing samples 189
 region play mode 97
Band-Pass Filter 146, 310
Bank
 & preset display 69
 create new from current multisetup 78
 create new using librarian 67
 description 32
 structure 57
Bank Select, MIDI 317
Bar/Beat 212
Basic MIDI Channel, definition 317
Batch Processing 202
Beats
 analysis 209
 button 116
 marker 212
 marker view 117
 parameters 213

- per bar changes 214
 - shifting right or left 213
 - synchronizing 222
 - tempo granularity 214
 - Bias, tube effect 266
 - Bit Depth 321
 - setting in Cubase LE 321
 - Bit Reduction 190
 - Body Size, filter 145
 - BPM
 - envelopes 134
 - function generator 141
 - LFOs 137
 - Brownian, function generator mode 142
 - Buffer Latency Setting, Cubase LE 321
 - Buffer Size 24
 - Buffers, sample 23
 - Build Preset Options 297
 - Bypass Effects 240
- C**
- Capture MIDI Controllers 82
 - Catalog Disks 66
 - Category
 - preset 43, 71
 - CC Window
 - keyboard 109
 - link 126
 - Channel Lag Processors 85
 - Channel Ramp Processor 86
 - Channel Swap 189
 - Channel Sync, function generator 141
 - Chicken Systems, Translator 319
 - Chorus
 - effect 246
 - voice 130
 - Chromatic, sample placement 295
 - Classic Response, volume 154
 - Clear Clip Indicators After 10 Seconds 29
 - Clear Library 66
 - Clipping
 - headroom 24
 - preventing 74
 - Clipping, Proteus VX 324
 - Clock
 - divisors 305
 - external MIDI 134, 137
 - master 74
 - modulation 304
 - Coarse Tune/Transpose Differences 129
 - Coarse Tuning, voice 129
 - Codec, to audition 65, 171
 - Comb Filter 146, 256
 - Combine Voices 120
 - Complete, envelope generator mode 134, 302
 - Compression
 - sample time 190
 - tube effect 266
 - Conserving Polyphony 165
 - using envelope complete mode 134, 302
 - Constant Time, pitch change 192
 - Context Sensitive Menus 278
 - Continuous Controller 314
 - changing loops using 219
 - cross-switch between voices 108
 - Control Panel, E-MU 321
 - Controller Knobs, MIDI keyboard 324
 - Controllers
 - #7 27
 - initial values 81
 - MIDI 314
 - MIDI #7 314
 - MIDI real-time 314
 - MIDI setup 28
 - modify the sound using 45
 - modulate FX using 242
 - modulation source 301
 - sensitivity 27
 - Convert
 - 24-bit to 16-bit 190
 - other files to Emulator X format 319
 - other files to Proteus X format 319
 - sample rate 195
 - stereo to mono 194
 - Convolution 193
 - Copy and Paste, multiple samples & presets 276
 - Copy Section 180
 - Cords 306
 - amount chart 158
 - lore 161
 - modulation 301
 - preset 87
 - voice modulation 156
 - Cords, connecting 63
 - CPU Cap 24
 - CPU Meter 77
 - Crossfade
 - positional 101
 - random 304
 - random cycle group 113
 - using cycle groups 113
 - xtractor setting 198
 - Crossfade Looping 204
 - how to use 208
 - Cubase LE
 - project window 320
 - setting up 320
 - Current Channel Indicator 77
 - Current Preset, changing 45
 - Curve
 - glide 131
 - velocity 27

- Cut Section 180
- Cut/Copy/Paste 274
 - multiple samples 184
 - sample 179
 - synth settings 274
 - voices 274
- Cycle Groups 111

- D**
- Damping, high frequency 243
- DC Filter, sample 188
- DC Offset 157
 - example 144, 308
- Decay Time, reverb 243
- Delay
 - LFO 136
 - voice 131
- Delete
 - folder 242
 - multisetup 79
 - region 286
 - sample 179
 - sample zone 121
 - unused samples 206
 - voices 120
- Destinations, Modulation 306
- Details Page 119
- Devices Menu, Cubase LE 320
- Diffusion, reverb 243
- Diode, modulation processor 307
- Direction 97
 - function generator 142
 - region loop 97
- DirectSound 23
- Distortion
 - tube effect 266
 - twin effect 267
- Divisors, clock 305
- Drag & Drop 270
 - operations list 328
- Duplicate, sample zone 121
- Dynamic Range Adjust 24

- E**
- Early Reflections Reverb, effect 244
- Early Reflections, description 243
- Early Reflections, effect 254
- Echo, creating 250
- Edit Menu, keyboard shortcuts 327
- Effect Types
 - 1-band para EQ 254
 - 4-band EQ 255
 - chorus 246
 - compressor 248
 - early reflections 254
 - early reflections reverb 244
 - flanger 256
 - growl 258
 - limiter 259
 - mono delays 250
 - phase shifter 260
 - pitch shifter 261
 - reverb 244
 - reverb lite 245
 - ring modulator 262
 - SP12-ulator 263
 - tremulator bpm 264
 - tube 266
 - twin 267
- Effects
 - block diagram 235
 - bypass 240
 - creating robot voice 250
 - descriptions 243
 - enable/disable 237
 - list of 243
 - post FX sends 241
 - preset, adding 233
 - selecting preset 73
 - sends 75
 - typical usage 236
 - voice controlled amount 239
- EIII/EIV, converting files to Emulator X 319
- Eliminate Audio from a Mix 196
- Embedded Info 295
- Emulator IV banks, loading 41, 319
- End Step, function generator 141
- Envelope Follower, ring modulator 262
- Envelope Generators 301
 - description 132, 302
 - mode
 - complete 134
 - repeating 133
 - rates, scaling time of 134, 159
 - retriggering 302
- Envelope, ADSR programming 133
- Envelope, reverberation 243
- EOS, banks loaded from floppy disk 319
- Equal Power Mix, paste sample 182
- Equal Power, crossfade loop 204
- Equal Temperment Tuning 83
- Expand Tree Items When Selected 29
- Expand Voice 121
- Export
 - bank 328, 329
 - multisetup as new bank 78
 - sample 57, 171
 - sample, batch process 202
 - tempo map 171, 172
- Expression, MIDI 157
- External Audio Editor 30

External MIDI Clock 27, 74
 External Tempo Source 27, 74
 Extractor 196

F

Fade
 bars in voices and zones windows 100
 sample 186
 Figure Eight, reverb pattern 244, 245
 File Menu, keyboard shortcuts 327
 Filter 310
 bandpass 146, 310
 comb 146
 cost on voice count 23
 descriptions 145
 FX
 1-band para eq 254
 4-band eq 255
 global type 77
 morph designer 149
 notch 310
 parametric 312
 poles 311
 swept EQ 312
 types 310
 xtractor 197
 Z-plane 313
 Find, files using the librarian 67
 Fine Tuning 130
 linked preset 125
 Fingered Glide 166
 First Key, modulation 86
 Fixed Pitch Mode 130
 Flanger 256
 Flip-Flop 307
 Floating Keyboard 270
 Floppy disk, loading EOS banks from 319
 Focus Width, xtractor 197
 Footswitch 301
 Forward, function generator mode 142
 Forward, region play mode 97
 Freerun, function generator 141
 Frequency Range of Common Instruments Chart 333
 Frequency, LFO 136
 Function Generator 139, 301, 305
 FX Edit Screen 242
 FX Insert Chains 242

G

Gain Adjust
 sample edit tool 187
 using xtractor tool 198
 Gain Fade, sample 186
 Gain, filter 145
 Gain, xtractor 197

Gamelan Tuning 83
 Gate, modulation source 300
 Gate/Chop Controls 284
 Gating, samples 284
 Generator
 envelope 301, 302
 noise 301
 GigaStudio, converting files to Emulator X 319
 Glide 131
 controlling with function generator 143
 modulation source 300
 rate 131
 Global Controls 74
 Gosset-Smith Tuning 192
 Grid, function generator 139
 Groove Loops 205
 Group 93
 make new from selected voices 93
 Group N 93
 Growl, effect 258, 259

H

HALion, converting files to Emulator X 319
 Hard Drive Requirements 340
 Hass-Effect Panning 130
 Headroom/Boost Control 24
 Hi Morph Gain 151
 High Frequency Damping, stereo reverb 243
 High Hat Simulation 164
 High-Pass Filter 310
 Hold, gate/chop control 284

I

IDE Disk Drive Tips 340
 Implementation Chart, MIDI 334
 Import Sample 171
 audio file in Windows 328
 Emulator X bank samples 328
 WAVE, AIFF, REX2, MP3 270
 Importing, tempo track to a sequencer 173
 Impulse Response, using transform multiply 193
 Independent, reverb pattern 244, 245
 Initial Controllers A-P 81
 Input Source, sampling 283
 Inputs, adding to Cubase LE 321
 IntelliEdit 26, 273
 MIDI audition 185
 Intensity, transform multiply 194
 Inter-Aural Time Delay 130
 Inverting, modulation sources 303
 Isolate, specific instrument from a mix 196
 ITD 130

J

Just C Tuning 83
 description 84
Just Intonation 84

K

Key

 CC Win 85, 109
 glide, modulation source 300
 ranges & fades, adjusting 272
 transpose 129
 transpose, voice 128
 velocity, modulation source 300

Key CC Win, changing loop with 222

Key CC Window 157

Key Sync

 function generator 141
 LFO 137

Key Timer, poly 86

Key Window

 link 125
 overview 99

Key/Velocity Windows, exploring 50

Keyboard

 CC window 109
 display, in voices & zones 94
 key, modulation source 300
 mini, floating 270
 modes 166
 pressure modulation source 300
 ranges 99
 tuning tables 83

Keyboard CC Window 109

Keyboard Range
 setting 61

Keyboard Shortcuts 327

Keyboard, placing samples on 60

Kirnberger Tuning 83

Knob Tracking 29

L

Lag Amount, preset lag 85

Lag Processors

 preset 85
 voice 138, 307

Large Icons 276

Latch Mode 167

Latch, floating keyboard 270

Layering

 presets, using links 124
 voices & samples 101

Length Modulation, function generator 143

Level, gate/chop control 285

LFO 301

 delay 136
 flanger 256, 257
 sync 304
 tips & tricks 138
 variation 136
 waveforms 136, 303

LFO, connecting to the filter 63

Librarian 66

Librarian, using 58

Limiter 74

Limiting Polyphony 165

Linear Mix, paste sample 182

Linear, crossfade loop 204

Links 123

 new 123, 124
 preset 123
 subsume 124
 utilities 124

List of Presets & Samples 276

List View 41, 276

Lo Morph Gain 151

Load

 preset 64
 samples 58

Load Emulator 4 bank 41

Load Last Bank at Startup 29

Lock Loop 205

Lock Oscillators To Tempo 222

Lock Region 286

Loop 203

 autocorrelation 178
 backward 97
 button 205
 compression 204
 crossfade 204
 envelopes 133
 forward 97
 how to loop a sample 207
 length, locking 205
 marker view 117
 modulation 161
 processing 203
 select 205
 settings 205
 switching 222

Loop (n) Knob 217

Loop in Release 205

Loop Select (Continuous) 220

Loop Select (Jump) 221

Loops

 and multisampled voices 223
 changing using a continuous controller 219
 creating from analyzed sample 215
 default behavior 217
 different for each voice 226

- playing back 217
- synchronizing 222
- tips for making 216
- Low Frequency Oscillator 136, 301, 303
- Low Pass Filter 310

M

- Main Output Routing 237
- Marker 212
 - snap to 205
- Mash Up Processing 199
- Master Clock 74
- Master Transpose 74
- Master Tune 74
- Master Volume Control 74
- Matrix, reverb pattern 244, 245
- Melody (last) 166
- Merge, library object into the bank 67
- Meter, beat analysis 211
- Meters
 - main output 74
 - sample input 283
- MIDI
 - audition 185
 - bank select commands 317
 - cc cross-switch voices 108
 - cc#7 curve 27
 - channel in multisetup 72
 - channel, current 44, 45, 71, 72
 - clock, using external 134, 137
 - continuous controller 314
 - continuous controller setup 28
 - controller #7 27
 - controllers
 - global page 77
 - initial settings 81
 - controlling effects with 242
 - file generated from region xplode 117
 - implementation chart 334
 - master volume sysex 334
 - modes, definition 317
 - multi mode 317
 - note-on 131
 - omni mode 317
 - poly mode 317
 - preferences 26
 - program change enable/disable 27
 - program changes 317
 - real-time controllers 314
 - selecting input ports 26
 - velocity curve 27
- MIDI Connection, Cubase LE 322
- MIDI Response 27
- MIDI Track
 - recording in Cubase LE 325

- Mini Keyboard 324
- MiniMoog, solo mode 166
- Mix Analysis, extractor 200
- Mix/Tune Tab 99
- Mod Wheel
 - floating keyboard 270
 - modulation source 300
- Mode
 - fixed pitch 130
 - mono 339
 - multi 339
 - omni 339
 - poly 339
 - solo 166
- Modulation 299
 - cords 301
 - description 299
 - destinations 306
 - of effects 242
 - of loop 161
 - processors 307
 - source, random 304
 - sources 300
 - adding 161
- Monitor Input, sampling 283
- Mono Aftertouch 300
- Mono A-I 164
- Mono Delays 250
- Mono Mode 339
- Mono to Stereo Conversion 194
- Morph Designer 149
- Morphing Filters 149, 313
- MP3, import 171
- Multi Mode 317, 339
- Multi-core Processors 25
- Multiple Samples, processing 202
- Multiple Trigger 166
- Multisample Voices
 - description 37
 - viewing & editing 92
- Multisetup
 - basic operations 78
 - description 34
 - export as new bank 78
 - intro 42
 - MIDI channel 72
 - output assignment 72
 - saving 78
 - selecting 69
 - selecting presets 72
- Mute Voice(s) 96

N

Name
 key 94
 labels for continuous controllers 28
 multisetup 78
 preset 43
 region 285
 sample 162
 show preset 331
 sort presets by 49
 sort samples by 93
 template 279, 280

Network, searching with librarian 66

New
 preset 327
 sample zone 121
 sample, pasting as 183
 voice 120

New Features 17

No Snap 205

Noise Generator 301

None, in preset select field 45

Normalize
 sample 187
 xtractor 197

Normalize, sample 187

Not Set
 filter override 77
 twistaloop override 73

Notch Filter 310

Note Division, BPM envelopes 135, 137

Note Marker 212

Note Onset, marker view 117

Nudge Sample Rate 192

Nudge Tempo 224

Number of Samples Playing 77

Numeric Values, entering 271

O

Offset Sample Start 131

Omni Mode 317, 339

One-shot
 function generator mode 142
 loop mode 97

Open Sample 120

Open Voices 120

Optimizing your Computer 20

Original Key 60, 99
 setting in region list 286

Overview Section, voices & zones 95

P

Pan
 amplifer 154
 in multisetup 72
 linked preset 125
 selecting audio based on 196

Parametric Filter 312
 1-band para eq 254
 4-band eq 255

Paste Section 181
 as new sample 183

PatchCord
 amount 314
 amounts when routing to pitch 158
 description 301
 sources & destinations 156
 tutorial 52

PatchMix DSP, setting up for sampling 282

Pedal, modulation source 300

Pendulum, function generator mode 142

Phase Shifter, effect 260

Phase Shifter, filter 146

Phattening, using chorus 246

Ping Pong 252, 253

Pink Noise 308

Pitch Amount Chart 158

Pitch Change
 sample edit tool 192
 using xtractor tool 197

Pitch Detect, sample 295

Pitch Shifter
 effect 261
 xtractor 197

Pitch Wheel 300
 bend range 129
 control assignment 28
 floating keyboard 270

Play Button, sampling 283

Polarity, modulation source 160

Poly 16 A-B, assign group 164, 165

Poly All, assign group 164, 165

Poly Key Assign Groups 165

Poly Key Timer 86

Poly Mode, MIDI 317, 339

Poly Normal, keyboard mode 166

Poly Release Trigger 167

Polyphony, optimizing 23, 165

Positional Crossfade 101

Post FX Send Amounts 241

Pre/Post, gate/chop controls 284

Pre-Delay 243, 244, 245

Preferences 21

Pre-Filter 258

Pre-Roll 22
 RAM indicator 77

Presence 266, 267

Q

Preset

- about 34
- auditioning 45
- changing 43, 45, 70
- cords 87
- creating from samples 58
- description 34
- editing 80, 299
- links 123
- opening 49
- selecting from the tree 45

Preset Effects

- adding 233
- enable/disable 73
- modulation 242
- selecting 73, 241
- voice controlled 239

Preset Global Page 80

Preset Modulators 85

Processor

- channel lag 85
- channel ramp 86
- modulation 307
 - lag 308
 - quantizer 308

Program Change Commands 317

Program Change, Cubase LE 325

Project Window, Cubase LE 320

Proteus X Composer FX Template 237

Pulse Width Modulation 161

Q

Q 147, 311

Quad-core Processors 25

Quantizer

- effect 263
- modulation processor 307
 - example 308

R

Ramp, channel 86

Random

- crossfade 304
 - using cycle groups 113
- crossswitch between voices 106, 108
- function generator mode 142
- generator 301
- modulation 304
- voice assignment 111

Range, keyboard 99

Rate

- lag processor 138
- LFO (frequency) 136
- modulation, function generator 143
- step rate, function generator 141

Realtime Controller, tutorial 314

Realtime Window 105

Receive Program Changes 27

Record Button, sampling 283

Recording a MIDI Track, Cubase LE 322

Recording Audio 282

Recording Directory 30

Reducing Bits

- effect 263
- of a sample 190

Region(s) 226

- list 285
- page 115
- window 115
- xplode 117

Release Velocity, modulation source 300

Re-order Voices & Samples 274

Repeat, envelope generator mode 133

Resampling 293

Resonance

- controlling in realtime 159
- definition 311
- filter 147

Response, amplifier 154

Rest, creating with function generator 144

Restore, multisetup 78

Retrigger

- envelope 302
- function generator 143
- LFO 305

Reverb Lite, effect 245

Reverb, effect 244

Reverb, envelope 243

Reverberation, description 243

Reverse

- function generator mode 142
- sample 189

REX 2, import 171

Right Click Shortcuts 328

Ring Modulator, effect 262

Robot Voice Effects, creating 247

Rotational Voice Assignment 111

S

Sample

- audition from disk 65
- batch processing 202
- buffers 23
- calculator 195
- change pitch of 192
- cut, copy, paste multiple 184
- DC filter 188
- definition 35, 38
- deleting unused 206
- editing tools 185
- editor 169

- fade 186
- layering 101
- loading one into the bank 58
- locating 58
- normalize 187
- open 120
- output of Emulator X 293
- pasting as new 183
- placing on the keyboard 60
- pool 57
- retrigger 159
- reverse 189
- select all 205
- selecting multiple 58
- start offset 131
- truncation 186
- Sample Edit Menu, keyboard shortcuts 327
- Sample Rate
 - convert, sample 195, 196
 - nudge 192
 - setting 24
- Sample Zone
 - delete 121
 - duplicate 121
 - new 121
- Save 32, 57
 - controller settings 46
 - definition 32
 - FX & multiseups 78
 - template 279
- Save As 32, 57
- Save Text Listing 276
- Scarlatti Tuning 83
- Scribble Strip 77
- Section, morph designer 151
- Select All, sample 205
- Select Loop 205
- Select on Current Channel 72
- Select Preset on Active Channel 29
- Selecting Voices, using MIDI keyboard 273
- Send Aux 1-3 159
- Send Main 159
- Sensitivity, beat 116, 212
- Sensitivity, MIDI controller #7 27
- Sequencer, using function generator as 140
- Set Keys
 - sample acquire 286
 - sample placement 295
- Set Names
 - region 287
 - sample acquire 286
- Set Original Key, sample placement 295
- Shape
 - LFO 136
 - morph designer 151
- Shifting Beats 213
- Shortcuts, keyboard 327
- Show All Button, function generator 140
- Show Bank & Program Numbers 277
- Show/Hide Object Numbers 277
- Signal Path, simplified 127
- Single Cycle Loop 195
- Small Icons 276
- Smooth, function generator 141
- Snap
 - off 205
 - settings and markers 117
 - to marker 205
 - to zero crossings 205, 206
- Software Activation 19
- Software Installation 18
- Solo Release Trig 167
- Solo Voice(s) 96
- Solo, keyboard modes 166
- Sort
 - voices 93, 121
 - voices and zones 121
 - zones 121
- SoundFont, converting files to Emulator X 319
- Sources, Modulation 300
- SP12-ulator, effect 263, 264
- Start At Loop Button 217
- Step Rate, function generator 141
- Stereo to Mono Conversion 194
- Stereo Width, voice chorus 130
- Stop Button, sampling 283
- Streaming, description 22
- Subsume Links 124
- Summing Amps 307
 - using lag processors 138
- Summing Nodes 161
- Swap FXA & FXB 242
- Swap L-R Channels 189
- Sweep Control, xtractor 198
- Swept EQ Filter 312
- Switch, modulation processor 307
- Switching Voices
 - by cycling 111
 - by keyboard key 109
 - by velocity 103
 - randomly 108
 - using a realtime controller 105
- Sync
 - function generator 141
 - LFO 137
 - to clock 304
- Synth Solo Mode 166
- Synthesizer Controls 127
- SynthSwipe 290
 - parameters 292
 - setting up 290
- System Exclusive, master volume 74
- System Requirements 18

T

- T'switch 157
- T'switch FF 157
- Tempered Tuning 84
- Templates 87, 128, 279
 - creating and recalling 279
 - organizing 280
- Tempo
 - control 74
 - enable/disable 27
 - via preset 82
 - external control 27
 - fitting a sample to 190
 - granularity 214
 - nudging 224
 - range (bpm) 210
 - variation 210
- Tempo Map, export 171
- Text Listing of Presets & Samples 276
- Time Compression, sample 190
- Toolbars 269
 - viewing & hiding 270
- Transform Multiply 193
- Translator Application 319
- Transport Menu, keyboard shortcuts 327
- Transport/Loop Controls 177
- Transpose
 - floating keyboard 270
 - key 129
 - linked preset 125
 - master 74
 - non-transpose mode 130
 - preset global 81
- Tree
 - exploring 41
 - navigation using keyboard shortcuts 327
 - view 277
- Trigger, multiple keyboard mode 166
- Triggering Envelopes, from modulation sources 302
- Triplet Button 212
- Truncate
 - after loop 204
 - sample 186
- Tube, effect 266
- Tuning
 - coarse 129
 - keyboard 83
 - to other instruments 74
 - voice fine 130
- Tuning Tables 83
- Tutorials
 - Adding an aux effect 232
 - Adding an effect to a preset 233
 - Advanced application guide 65, 299
 - Building a preset from the ground up 58
 - Change loops with a controller 220, 221
 - Conserving polyphony 165
 - Control FX parameters via MIDI 242
 - Control the effect amount from the voice 239
 - Create a custom bank 47
 - Creating ADSR envelopes 133
 - Crossfade between two loops 227
 - Cycle groups for more realistic instruments 112
 - Examine a voice 52
 - Examine the samples 54
 - Function generator as an arpeggiator 140
 - Importing a tempo track to a sequencer 173
 - Layering presets 124
 - Looping a sample 207
 - Morph designer 150
 - Opening a preset 49
 - Process a vocal and return it to the mix 199
 - Program change commands 317
 - Proteus VX with Cubase LE 322
 - Pulse-width modulation 161
 - Sampling 282
 - Setting up your controller knobs 314
 - SynthSwipe 290
 - Using DSP 55
- Twin, effect 267
- TwistaLoop 209
 - bar numbering changes 214
 - override 73
 - troubleshooting 228

U

- Ultra-High Precision Interpolation 24
- Undo & Redo 32, 179
- Unused Samples, delete 206
- Update Library 58, 66
- Use Multiple Processors 25
- Use Optimal Buffer Size 24
- Utilities, link 124

V

- V2, what's new? 17
- Vallotti Tuning 83
- Variation, LFO 136
- Velocity Curve, MIDI 27
- Velocity Window 103
 - link 125
- View Menu 275
- Vista, audition sample under 65
- Vocal Elimination 196
- Voice 51
 - & sample zones 89
 - add to preset 51
 - assign group 164
 - coarse tuning 129
 - combine 120
 - continuous controller cross-switch 108
 - control of FX amounts 239

- create new voices from regions 117
- crossfading by realtime controller 105
- crossfading by velocity 103
- delay 131
- delete 120
- description 34, 36
- expand 121
- fine tuning 130
- fixed pitch mode 130
- key transpose 129
- keyboard range 99
- lag processors 138
- layering 101
- new 120
- open 120
- place into a group 93
- processing, basics 62
- random crossswitch 106
- realtime crossfade 105
- realtime window 90, 105
- select all 93
- select group 93
- selecting multiple 127
- selection 91
- sorting 93
- switching by key position 100
- switching by realtime controller 105, 108
- switching by velocity 103
- utilities 120
- velocity range 90, 103
- Voice & Sample Zone Utilities 120
- Voice Editor, overview 128
- Voice Effect Sends 155
- Volume
 - changing relative 95
 - in multisetup 72
 - increasing overall 24
 - linked preset 125
 - preset global 81
- Volume Control, master 74
- Volume Curve 27
- Volume Sensitivity 27
- VST Connections, Cubase LE 321
- VST Instrument, Cubase LE 323
- VSTi
 - & multisetup 78
 - Cubase LE checklist 326
 - launching 320
- VU Meter 74
 - recording 282

W

- WAVE, import 171
- Waveforms, LFO 303
- Wavetable Scanning Synthesis 161
- Werkmeister Tuning 83
- Wet/Dry Mix, effects 240, 241
- Wheels, Pitch & Mod. 28, 300
- White Keys 295
- White Noise 304

X

- X2, what's new? 17
- X3, new features 17
- Xplode, voices from regions 117
- Xtractor 196

Z

- Zero Crossing 206, 208
 - snap to 206
- Zero Crossing Threshold, setting 206
- Zoom, keyboard shortcuts 327
- Z-plane Filter 149, 313

