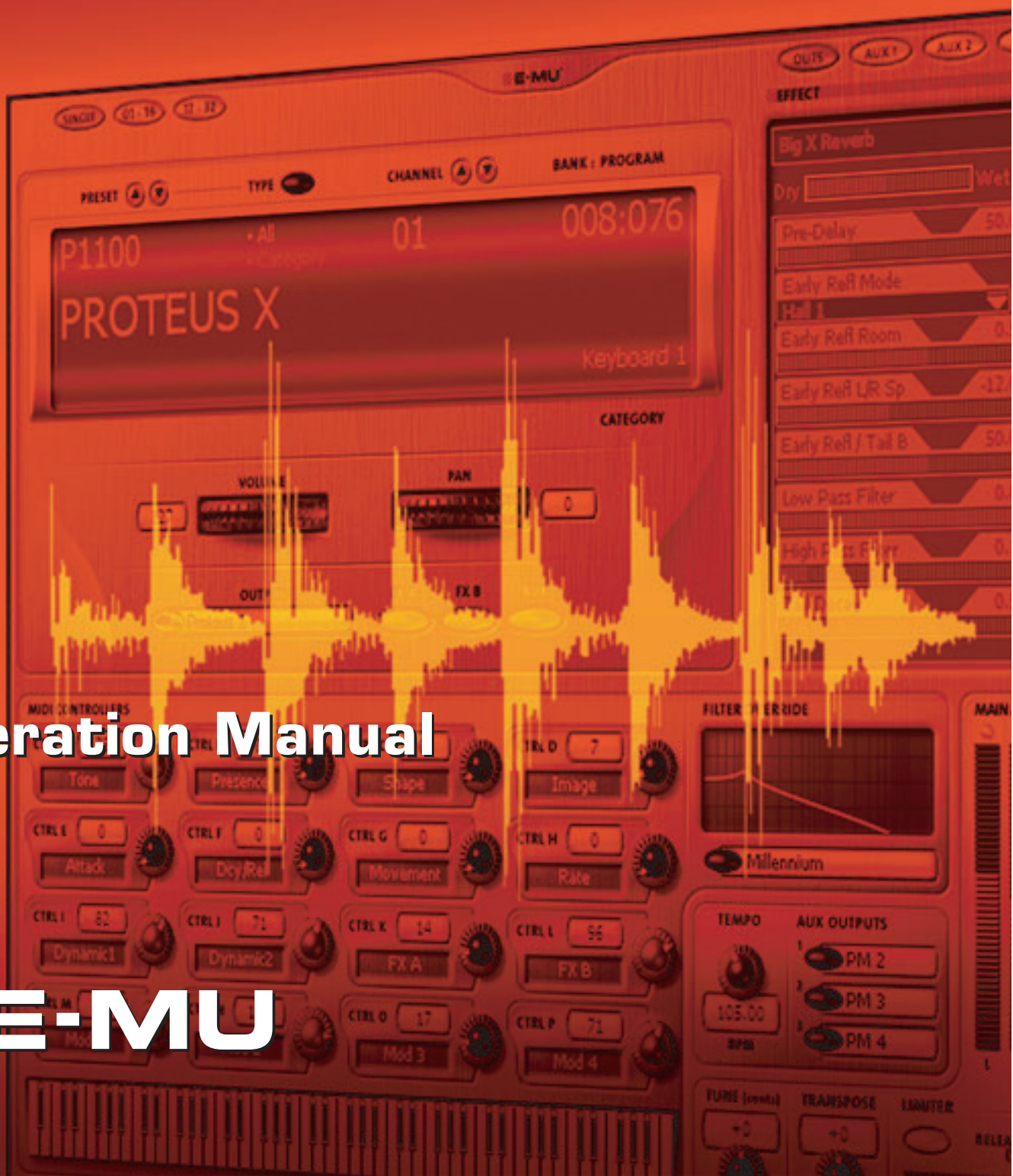
ProteusX

Desktop Sound Module



Operation Manual

E-MU

ProteusX

Desktop Sound Module

Operation Manual

Revision A

E-MU World Headquarters
E-MU Systems
1500 Green Hills Road
Scotts Valley, CA
95067-0015
USA

Europe, Africa, Middle East
Creative Labs
Ballycoolin Business Park
Blanchardstown
Dublin 15
IRELAND

E-MU Japan
Creative Media K K
Kanda Eight Bldg., 3F
4-6-7 Soto-Kanda
Chiyoda-ku, Tokyo 101-0021
JAPAN
www.japan.creative.com

Table of Contents

1 - Introduction & Installation

Introduction

The Proteus X is a professional desktop sound module based on the Proteus sound modules and Emulator samplers. The engineers and designers at E-MU Systems have brought their years of experience and expertise to create the most comprehensive and flexible software instrument ever.

Proteus X

Proteus X allows quick and easy creation of multitimbral setups on up to 32 different MIDI channels. Other features include the following:

- 24-bit playback
- Sample streaming from the hard disk allows huge sample banks.
- Up to 192 kHz sample rate
- 32-bit floating point internal processing
- Drag and drop sample, voice and preset selection
- Phase Locked Stereo signal path
- Ultra High Precision Pitch Interpolation (user selectable)
- High Voice Polyphony (system dependent)
- Real-time controllers for all important control points

Proteus X allows the serious synthesist and programmer to literally design custom instruments using the renowned Proteus 4 synthesis architecture.

- 54 different filter types including multi-pole resonant filters, phasers, flangers, vocal filters and multi-parameter morphing filters.
- Extremely powerful, yet easy to use voice and sample zones allow crossfading by position, velocity or real-time controller up to 128 layers deep.
- Three multi-stage envelopes, two lag processors and two multi-wave LFO's per voice.
- Multiple solo modes and assignment groups let you simulate the playing response of physical instruments and classic synthesizers.
- 36 patchcords per voice allow you to get as complex as you want when building instruments. Arithmetic modifiers in the patchcord section allow you to construct complex synthesis models.

Before you Begin...

You should have a good working knowledge of you 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.*

System Requirements

To use Proteus X you will need at least:

- Pentium III 1GHz or equivalent CPU (P4, 2 GHz or faster recommended)
- 512 MB RAM (1 GB DDR or greater recommended)
- Hard disk capable of 200 Mb/sec transfer rate.
- Windows 2000 or XP

Software Installation

Programs installed

- Proteus X Desktop Sound Module application
- Proteus X VSTi
- Proteus X Converter
- Proteus X Factory Sound Banks

Important: *Install the EMU Digital Audio System audio card before installing and running Proteus X.*

1. Insert the CD labeled "Proteus X Program Disc" into your CD-ROM drive.
2. On the CD-ROM, locate the file called "Install Proteus X" and double click on it.
3. Follow the instructions on the screen.
   - The Proteus X applications have been installed inside "Program Files/Creative Professional/ProteusX."
4. Look for any "ReadMe" files in the Proteus X folder or in the Start menu and read these before continuing. The readme files may contain important last minute information that didn't make it into this manual.
5. Remove the Proteus X application disk from the CD-ROM drive.

Install the Proteus X Sound Banks

6. Locate "Sound Disk 1" and insert it into the CD-ROM drive. This disk contains the **Proteus Composer** bank used in the tutorials.
7. If Windows AutoPlay mode is enabled for your CD-ROM drive, the installer begins running automatically.
8. Follow the onscreen instructions to install the bank on your hard disk. The Factory Sound Banks are installed inside "Program Files/Creative Professional\ ProteusX\Production Sound Banks" by default.
9. You can install the other sound bank disks now or at a later time.
10. Launch the Proteus X program by double clicking the Proteus X icon on the desktop or by selecting the Proteus X item from the Start menu.

Register your Software

Please register your software today to guarantee uninterrupted use.

Optimizing Your Computer for Proteus X

Windows: 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 Proteus 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 controller used by your audio drive.
- Get the fastest hard disk drive you can afford, as this is the single most important component affecting Proteus X performance. Many drives now offer an 8MB data buffer, 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 the System Restore feature of Windows XP 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 SCSI or IDE drive for use with Proteus 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 Proteus 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

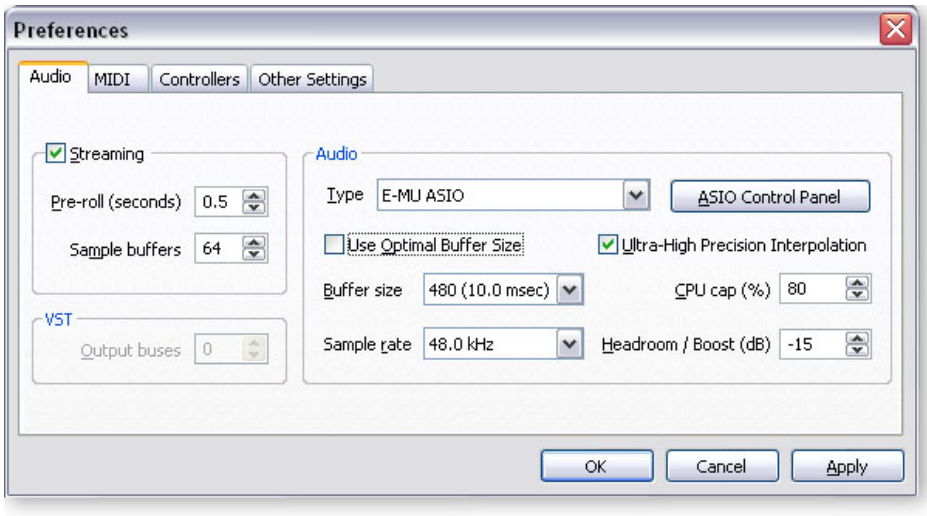
16 bits	44.1 k sample rate	86.2 k per Channel per second
	48 k sample rate	93.8 k per Channel per second
24 bits	44.1 k sample rate	138 k per Channel per second
	48 k sample rate	150 k per Channel per second

- 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.

Setting up your Audio

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

1. Select the **Preferences** screen from the menu bar.



2. In the Audio section of the dialog box, select **E-MU ASIO** (if not already selected). This setting interfaces with the E-MU Digital Audio System.
3. With E-MU ASIO selected, the Proteus X will interface properly with the E-MU Digital Audio System. After running Proteus X, you may want to adjust the Audio Preferences for optimum performance on your computer system.
See Audio Preferences on page 16.
4. If for some reason you want to use hardware that doesn't support ASIO, select the DirectSound driver. For Proteus 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.

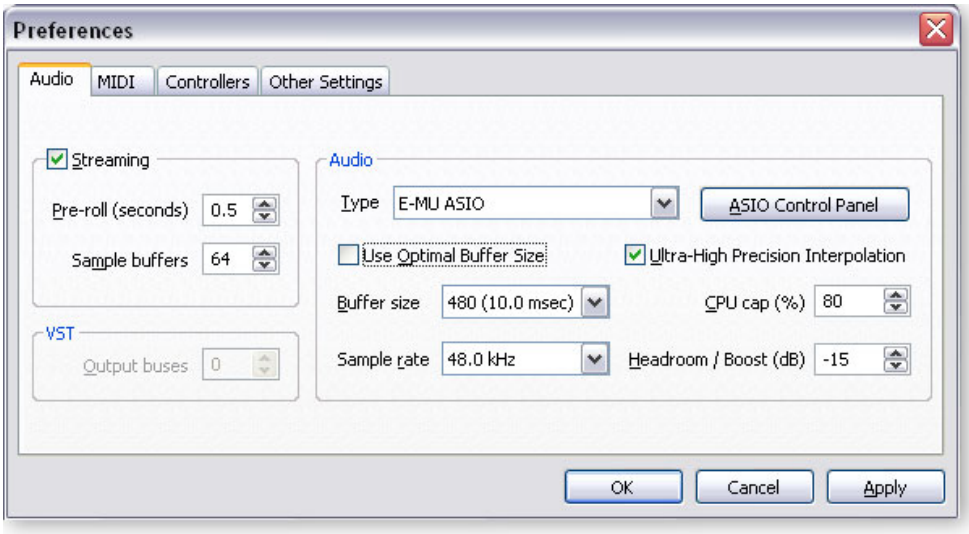
Please go to page 16 for complete explanations of the Audio Preferences.

Setting up the Preferences

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

▶ To Access the Preferences Dialog Box

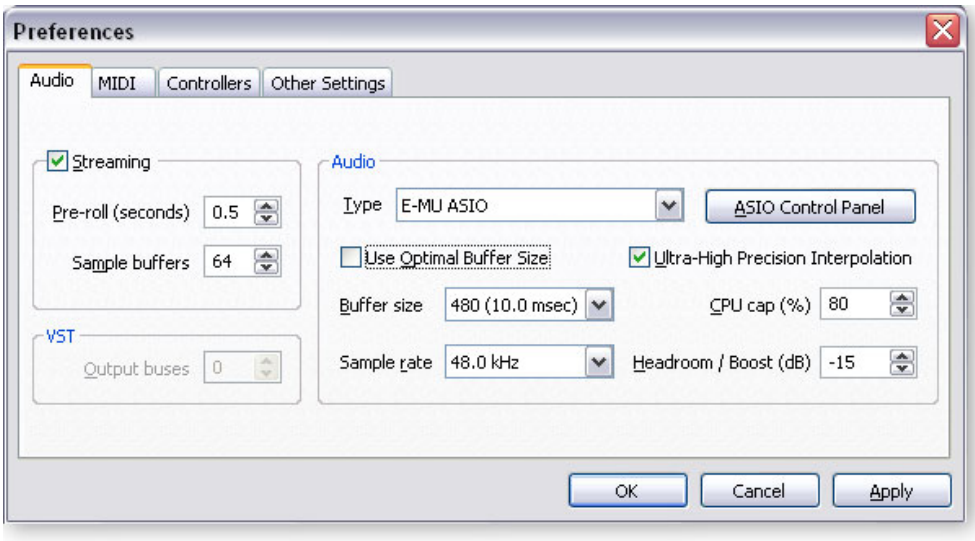
1. Select **Preferences** from the **Options** dialog box. The following screen appears.



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, Audio Hardware Setup & Pitch Shift Algorithm selection.
MIDI	MIDI Inputs Channels 1-16 & 17-32, IntelliEdit, and Receive/Ignore Program Changes, Internal/External Tempo Source select
Controllers	This section allows you to set up the MIDI Continuous Controllers you want the Proteus X to receive.
Other	This group contains three miscellaneous controls that didn't fit anywhere else: Load Last bank At Startup, Clear clip indicators after 10 seconds

Audio Preferences



Streaming

When you load a Proteus 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 the 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 sample player. Multi-layer, minute-long, stereo samples on each key are no problem for Proteus X. Sample streaming technology is not new. It was originally developed 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, the Proteus loads (or attempts to load) the entire bank into your computer's RAM.

▼ *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.*

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 Proteus 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

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 Proteus 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 Proteus X.

❖ **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.

The smaller the pre-roll, the more disk access will be required for sustained sounds. Frequent disk access can overtax your computer system depending on its speed and the number of other disk dependent applications you have running. Large pre-roll times will 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.

Sample Buffers

This control allocates the amount of RAM that will be set aside for Proteus 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 loaded 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.

▶ To Optimize the Performance and Polyphony of your System:

The polyphony of the Proteus 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 **bit depth** of your samples —16 or 24 bits (24 bits = fewer voices)
- The **sample rate** of the audio hardware—44.1kHz, 48kHz, 96kHz.
- The **speed of your computer**—CPU speed, RAM and hard disk access time.

Begin by choosing 44.1kHz or 48.kHz as your output sample rate.
The software-based filters can eat up CPU cycles and reduce polyphony.

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. (polyphony is divided by 2)
12th Order	Additional CPU load comparable to playing 2 more samples. (polyphony is divided by 3)

Synthesizer parameters also use CPU cycles. 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 Proteus 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. **To use Proteus X with the E-MU Digital Audio System, select the E-MU ASIO 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 speed and 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. The default setting is 128 sample periods, a good compromise of performance to CPU usage.

Use Optimal Buffer Size
When this box is checked, the Proteus 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 Proteus X, and will always match the sample rate of your sound card. The default setting is 44.1kHz.

CPU Cap
This control sets percentage of CPU resources that will be used by Proteus X and consequently the maximum number of samples that can be played at the same time. Depending on the design of the preset, a single Proteus X voice may contain multiple samples which 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 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 -15 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 Button
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, the Proteus X shifts the pitch of the available samples to fill in the keyboard.

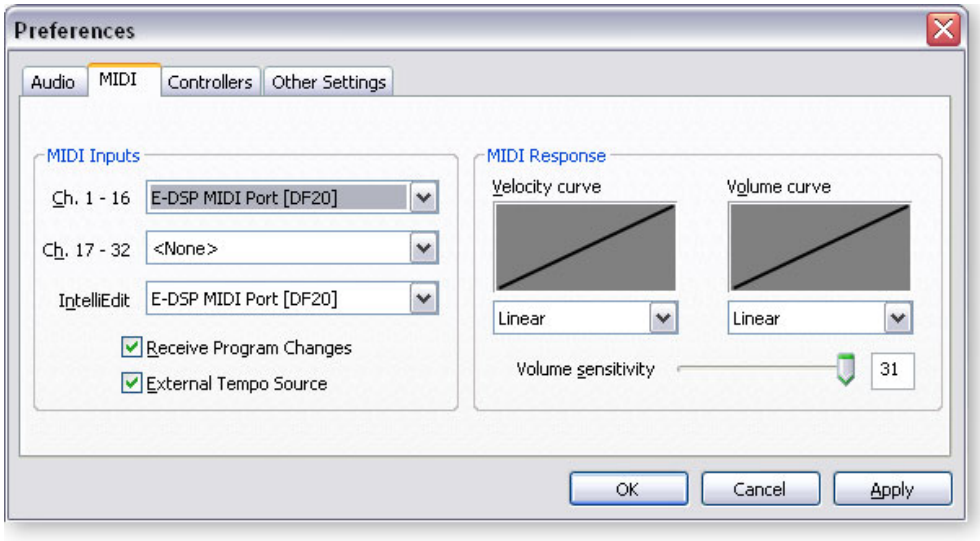
ASIO Control Panel
This button brings up the E-MU ASIO control panel which allows you to set the ASIO buffer latency in milliseconds. This important setting determines how fast notes will sound after you play them. If this time is too long, (>10mS) the keyboard response will feel slow and sluggish. If 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.

VST Output Buses

This control is only active in the Proteus X VSTi application. This setting allows you to add additional VST buses (stereo pairs). Proteus 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

The Proteus X can respond to 32 MIDI channels at once. Since there can be only 16 MIDI channels per MIDI cable (or input) two inputs are provided. Set the MIDI Inputs (Ch. 1-16, Ch. 17-32) to match up to two outputs from your MIDI interface or sequencer. If you don't need 32 channels you can turn either 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. Here's how it works.

▶ 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 62.

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 the Proteus 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. The Proteus X uses a global master tempo for Tempo-based Envelopes, Tempo-based LFOs, and clock modulation. 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.

▼ **Important!**
If External Tempo Source is set to external, the tempo control in the Voice Editor will be "greyed-out" and non-functional.

MIDI Response

These controls allow you to customize the MIDI response of the Proteus 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 the velocity data unaltered. The shape of the selected curve is displayed in the window. Select the curve that works best for you.

Volume Curve

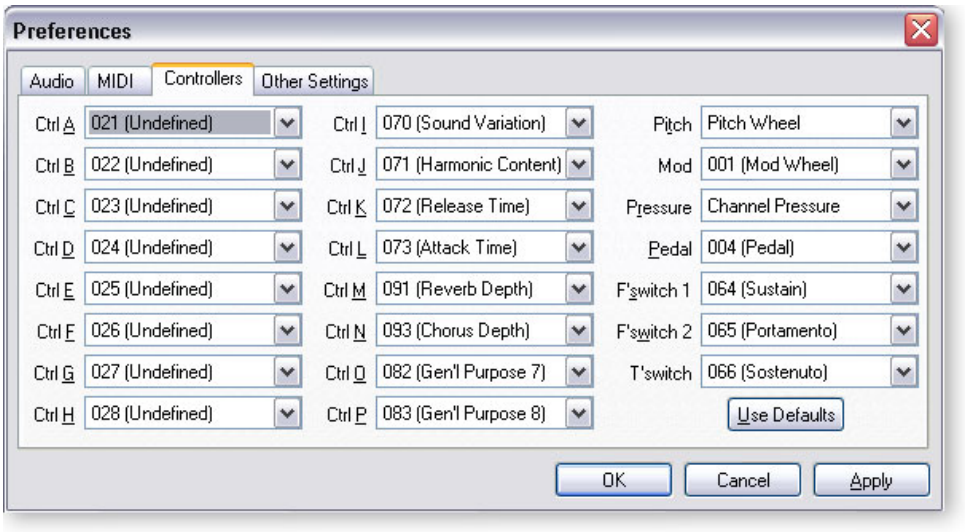
This is an adjustment to help match MIDI controller #7 volume response to other manufacturers' equipment. Three curves are provided: Linear, Inverse Square or Logarithmic. The action of this control is displayed in the window.

Volume Sensitivity

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

Controllers

This is where you set up which MIDI Continuous Controllers the Proteus 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 92.

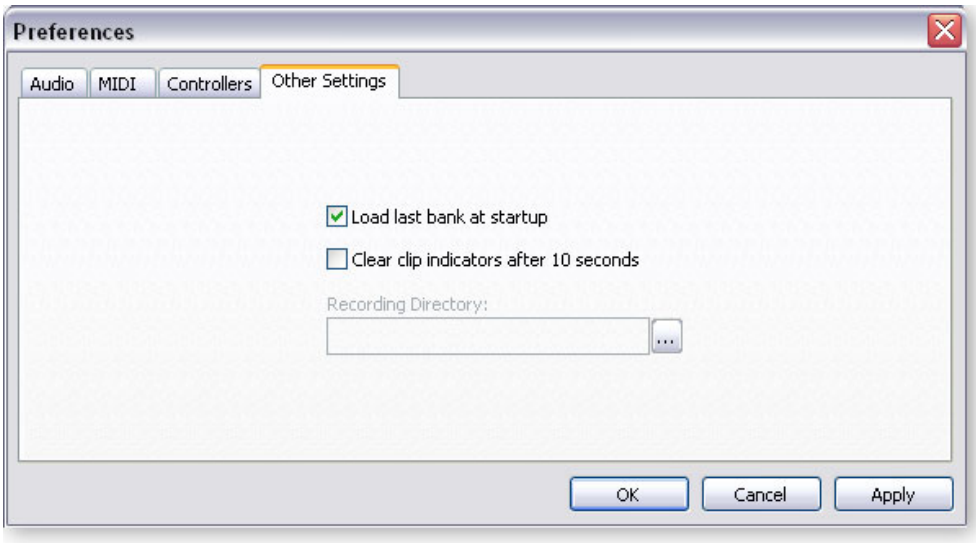


As you can see, MIDI Continuous Controller numbers are assigned to letters (A-P) or labels (Pitch, Mod, Pressure, Pedal, etc.) in this screen. When programming a preset, these letter or name labels can be assigned to control various parameters such as filter frequency or attack time.

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 which are shown above.

Other Settings



Load Last Bank at Startup - This feature automatically loads the last bank you had loaded whenever you start the Proteus X application.

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.

Appearance Settings in Windows

Adjusting the "Performance Options" in Windows will make the Preference buttons much easier to read and increase the smoothness when dragging windows around on 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**.

2 - Proteus Architecture

This chapter contains important background information about the organization of the various modules of the Proteus 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!

A Modular System

You can think of the Proteus 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 the Proteus X hierarchy starting from the largest element, the Bank.

Bank

Before you can play the Proteus 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, the Proteus X only loads a small portion of the digital samples so bank load time is greatly reduced.

Saving

The bank retains data only for as long as your computer is plugged in and turned on. Of course, we don't expect you to leave your computer on all the time, which brings us to the subject of saving data.

Saving the bank to a hard disk permanently stores data so that even after turning off your computer, the disk stores a record of your work.

IF YOU DO NOT SAVE A BANK, ANY CHANGES YOU HAVE MADE WILL BE LOST WHEN YOU TURN OFF THE COMPUTER!

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 erase the computer's memory. 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 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.

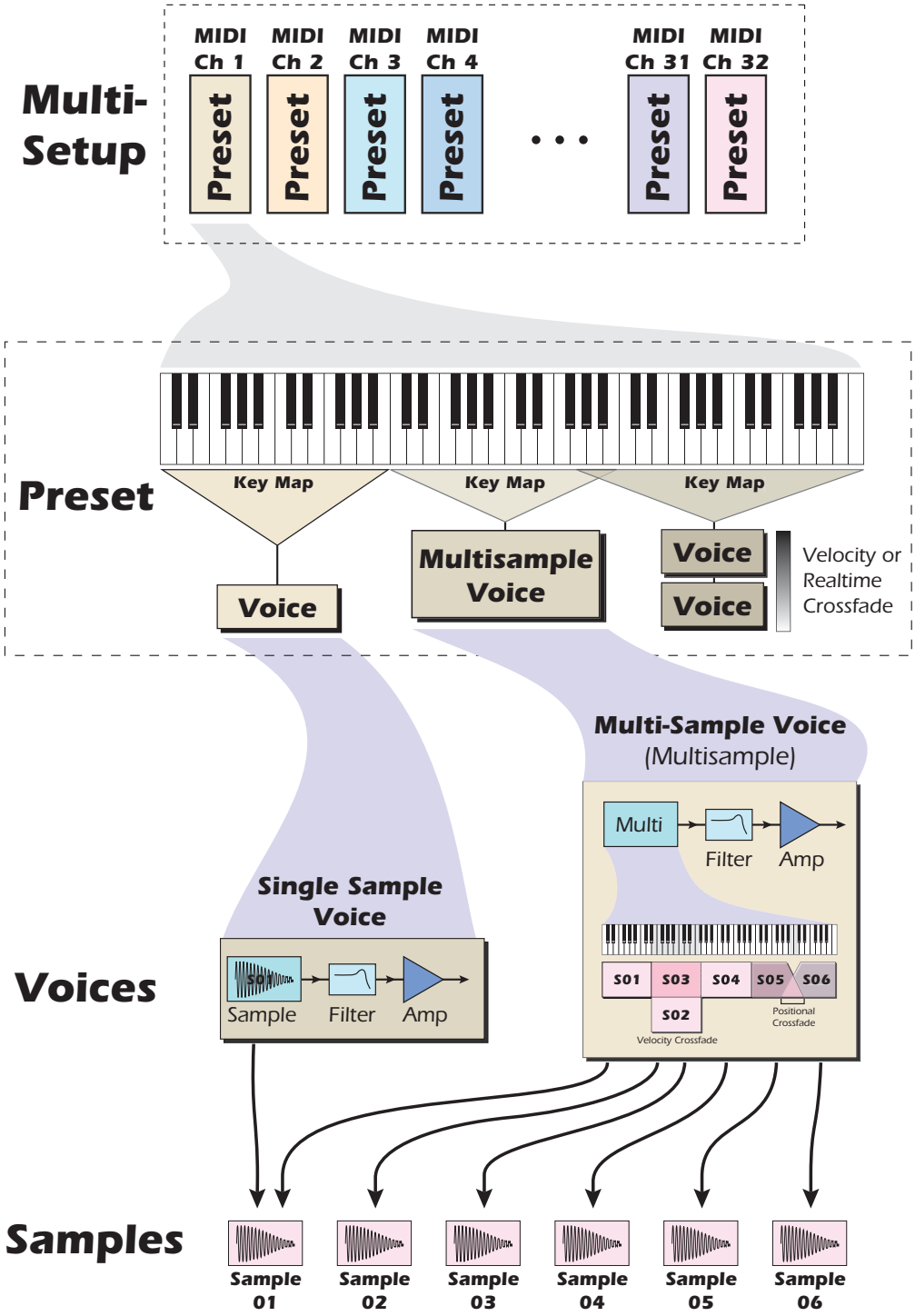
▼ Whenever you have done enough work that you would hate to lose it, **BACK IT UP!**

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

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

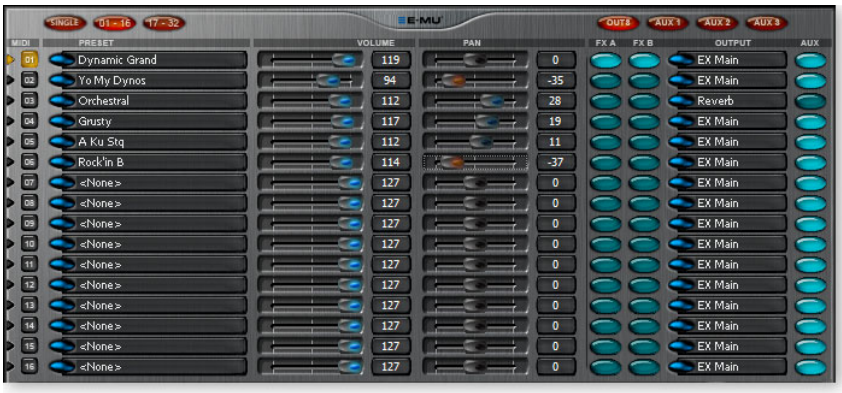
When you choose "**Save As**" from the file menu, the entire bank—presets, voices and samples—are re-written to the hard disk. This method, although somewhat wasteful of memory, ensures that all your samples remain bound with your bank.

Proteus X Bank



Multisetup

A Multisetup is like a map which assigns a preset, volume, pan position and output routing to each of the 32 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 loop data.

Flexible Architecture

The Proteus 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 be used to crossfade, layer or switch multiple complex voices.

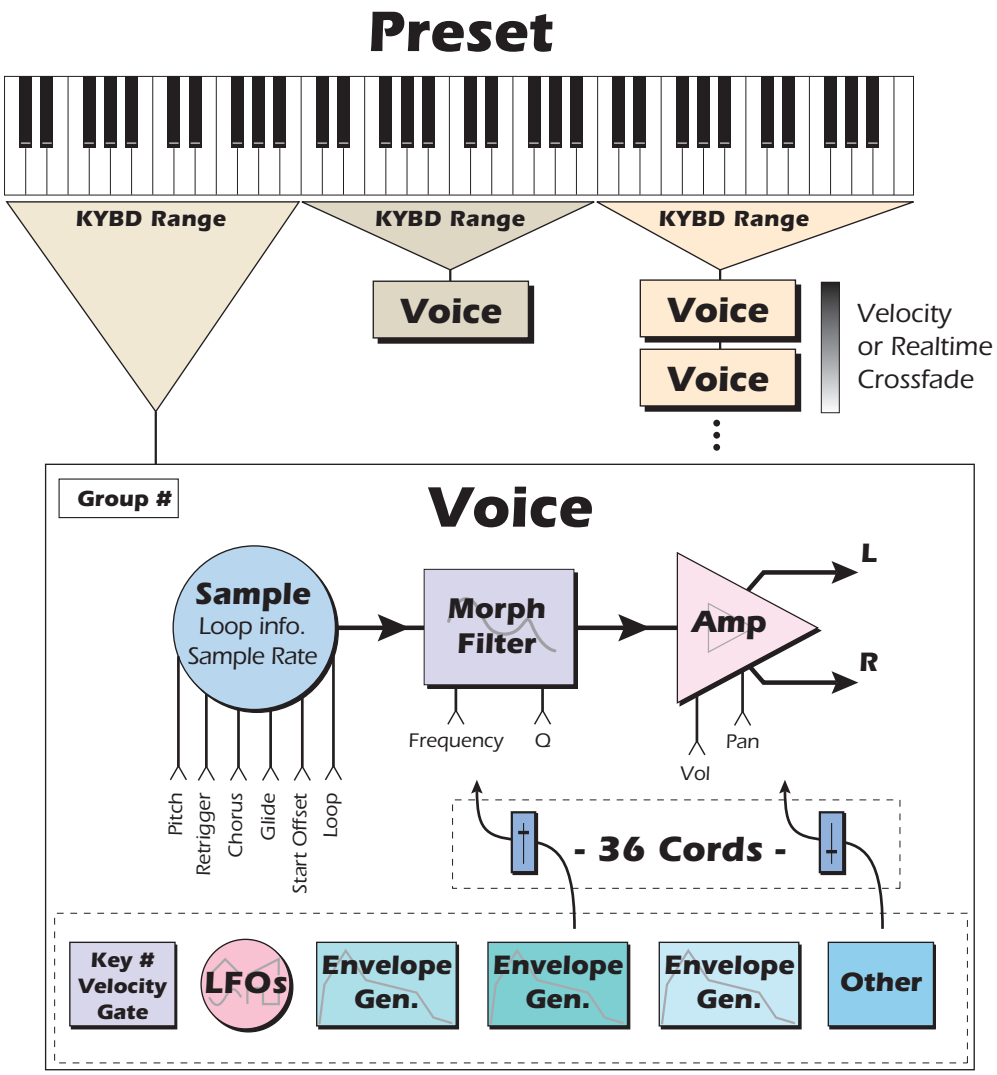
Preset Diagram

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

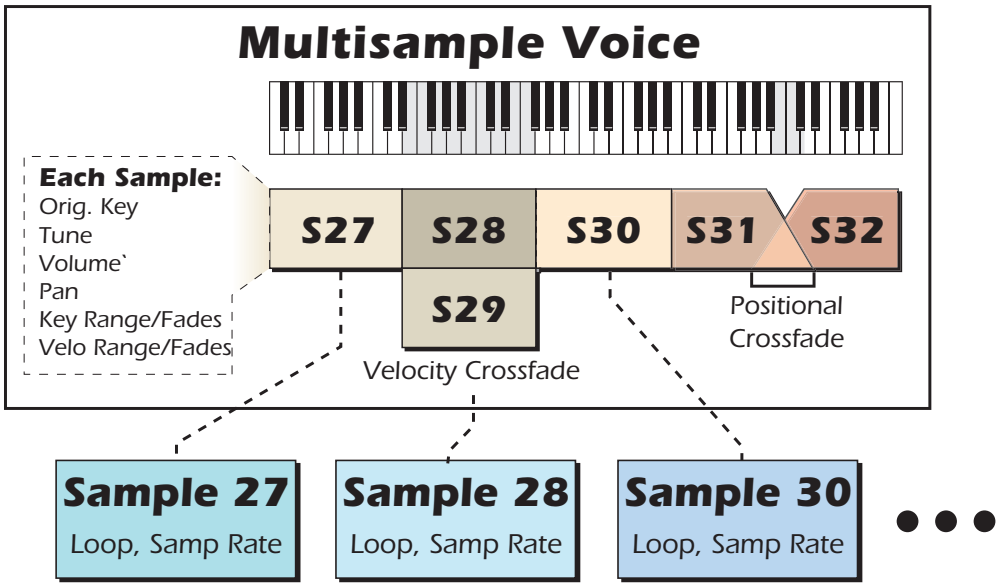
A voice is a complete sound that can be assigned to a range of the keyboard. A sample is the part of a voice that generates the actual sound. You can think of a voice as a complete instrument consisting of one or more samples, which can then be used as a building block in constructing more complicated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

A voice consists of one or more samples, a dynamic filter, a dynamic amplifier, up to three 6-stage envelope generators, up to two multi-wave LFOs and up to 32 modulation routings called "Cords" to connect everything together.

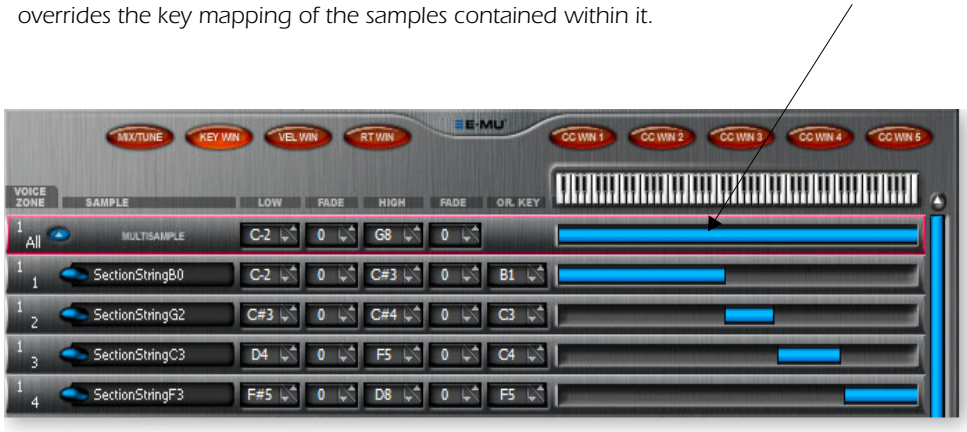
Multisample Voices

In another scenario, you might have 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 mapping of the multisample overrides the key mapping of the samples contained within it.



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."

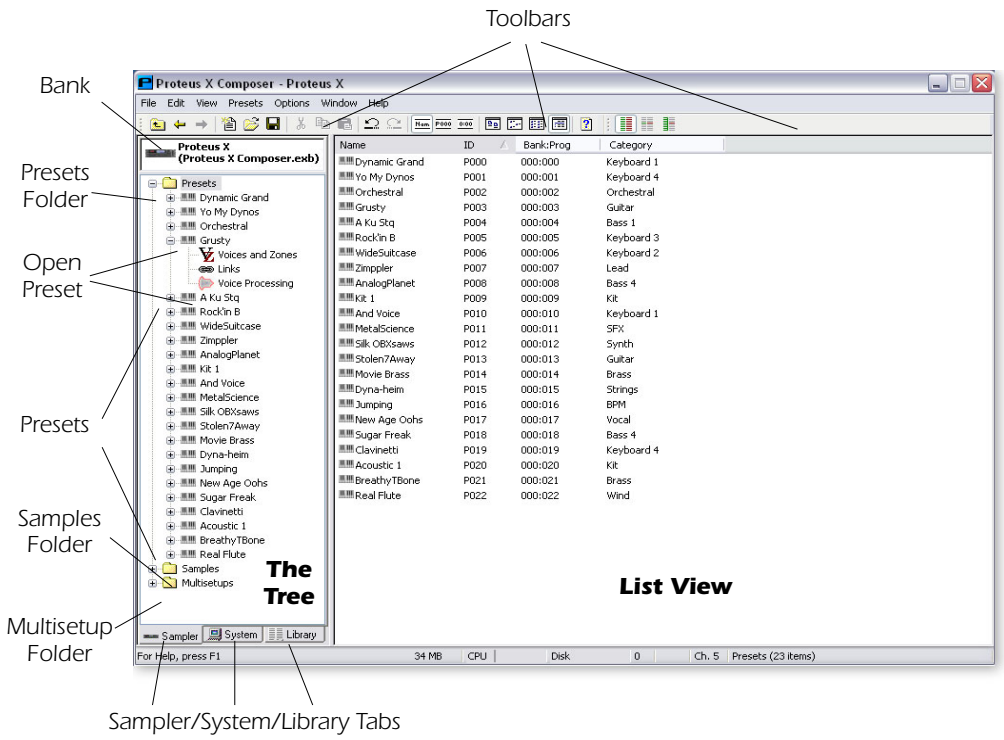
You can place the voice (and its sample) anywhere on the keyboard. If more than one voice is assigned to the same range, then pressing a key in that range plays all the voices assigned to that range. Voices assigned to the keyboard can be crossfaded by their position on the keyboard, or the key stroke 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.

3 - Getting Started

Exploring the Tree

In this section you'll learn how to navigate around Proteus 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 Proteus 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**.

▼ 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.

Sampler — Allows you to see the contents of the current Proteus 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 IV banks from this tab.***

Library — The Library shows you all Emulator X and Proteus X 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.

Multisetup Page

The Multisetup is the top level of the Proteus 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 32 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.

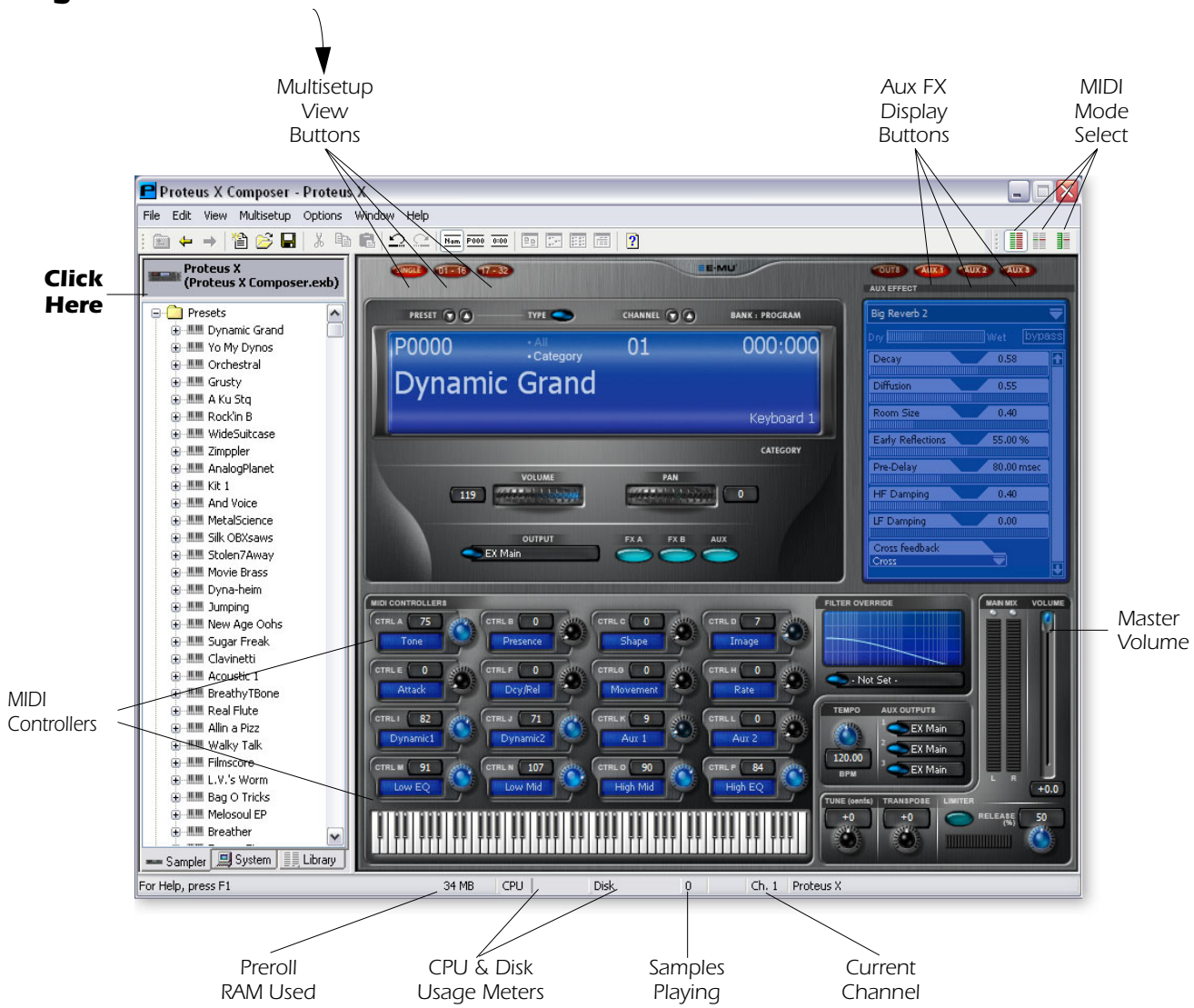
Click on the Proteus X icon to display the Multisetup page.

There are three 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.

See page 49 for a complete description of the Multisetup screen.

Single Channel View



Changing Presets (from Single View)

Single View is the default view when you first run Proteus 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

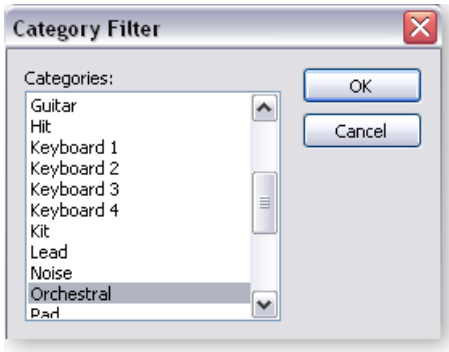
■ The Bank/Program Display shows the Bank and Program number you would use to select the preset from a sequencer.

▶ 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.



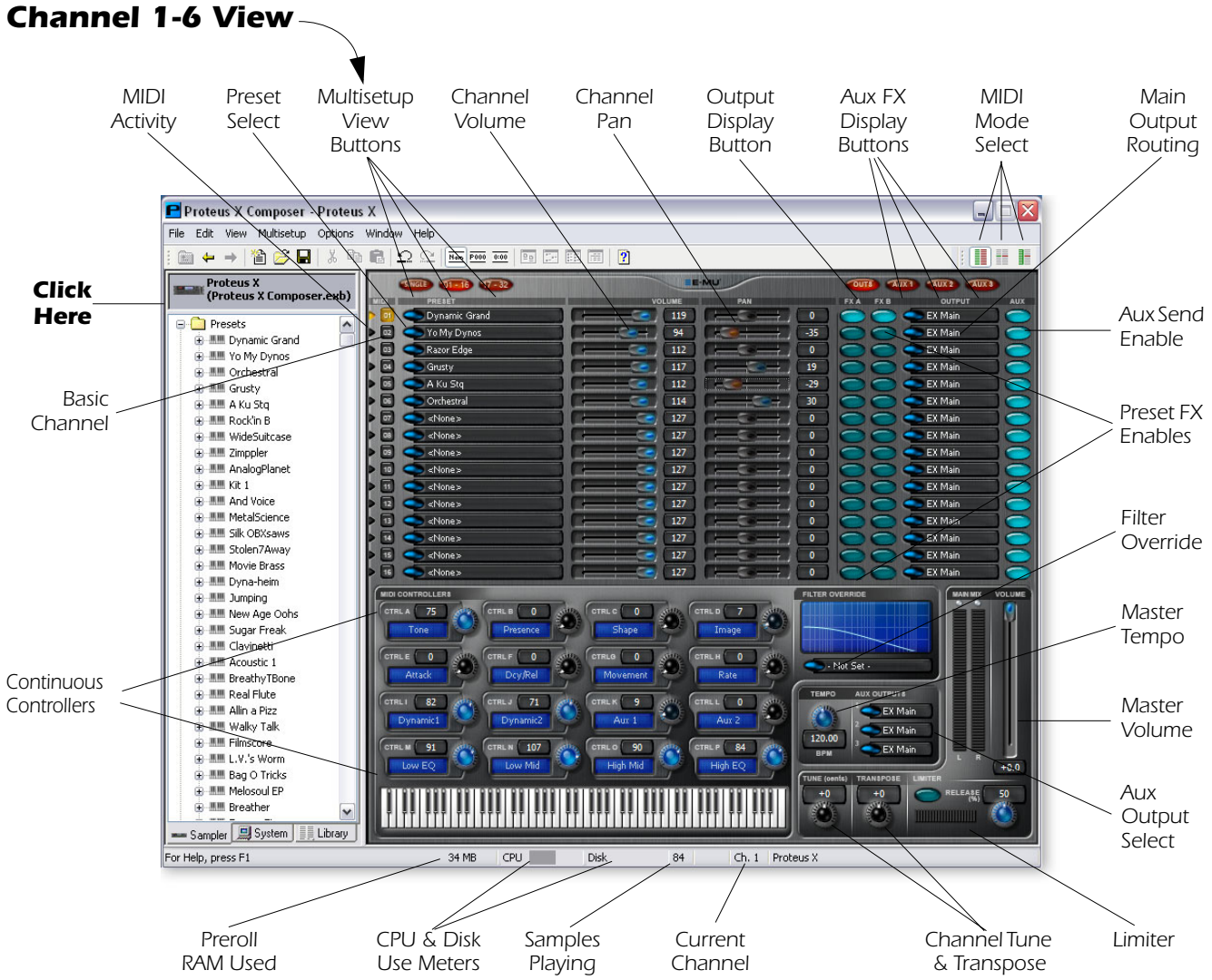
▶ Changing the Current MIDI Channel

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.

Channel 1-16 View

Press the Multisetup View 1-16 button 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. Press the 17-32 button to view the presets on channels 17-32. See also "4 - Multisetup / Global Controls" on page 49.

Channel 1-6 View



Changing Presets (from the 1-16 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 19.

▶ 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 preset by clicking on the number to the left of the preset. The number you selected turns red.
5. Play the small keyboard at the bottom of the screen. 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. The Proteus now plays the preset on that channel.

❖ Selecting "None" in the Preset field, disables the MIDI 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 **Proteus X icon** at the top of the tree.

▶ 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 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 the Proteus X if you match the CC numbers your MIDI keyboard transmits to the same CC numbers on the Proteus X. These global settings are located under Options, Preferences, Controllers. See page 21 and page 158 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 53 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.

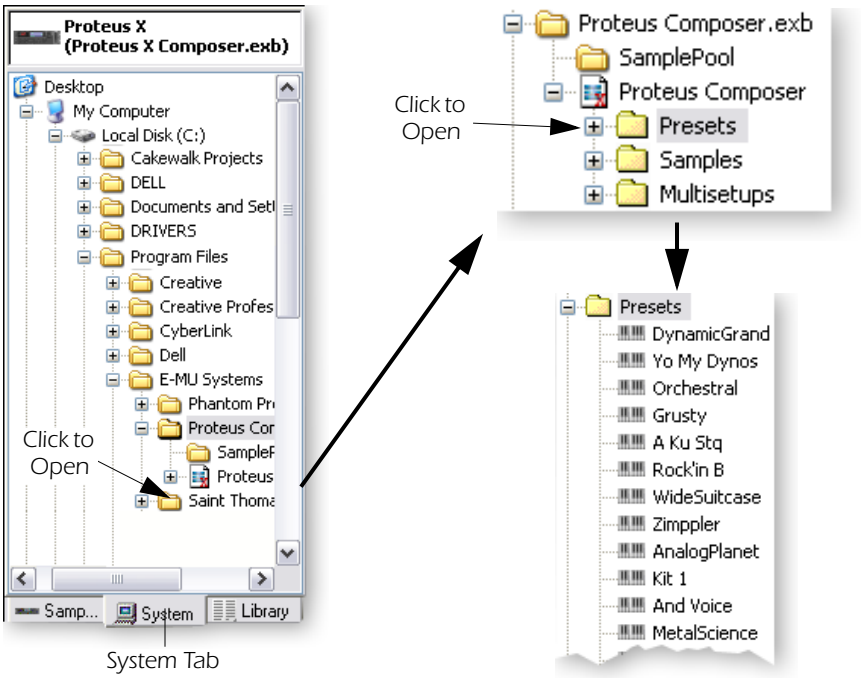
Create Your Own Custom Bank

It's easy to create your own custom sound banks by simply "dragging and dropping" the presets or samples you want onto the Proteus 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 "E-MU Systems" 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 **Proteus 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.

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 **Proteus 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 **Proteus X icon** at the top of the tree.

▶ 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 you just opened.
2. Click on the **Proteus X icon** in the Tree to view the Multisetup page.

Note: 24-bit samples are shown as 32-bit because they are internally stored and processed as 32-bit numbers.

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.

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 Proteus 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 **Cutlers**) to un-merge the sample.

Load a New Bank

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

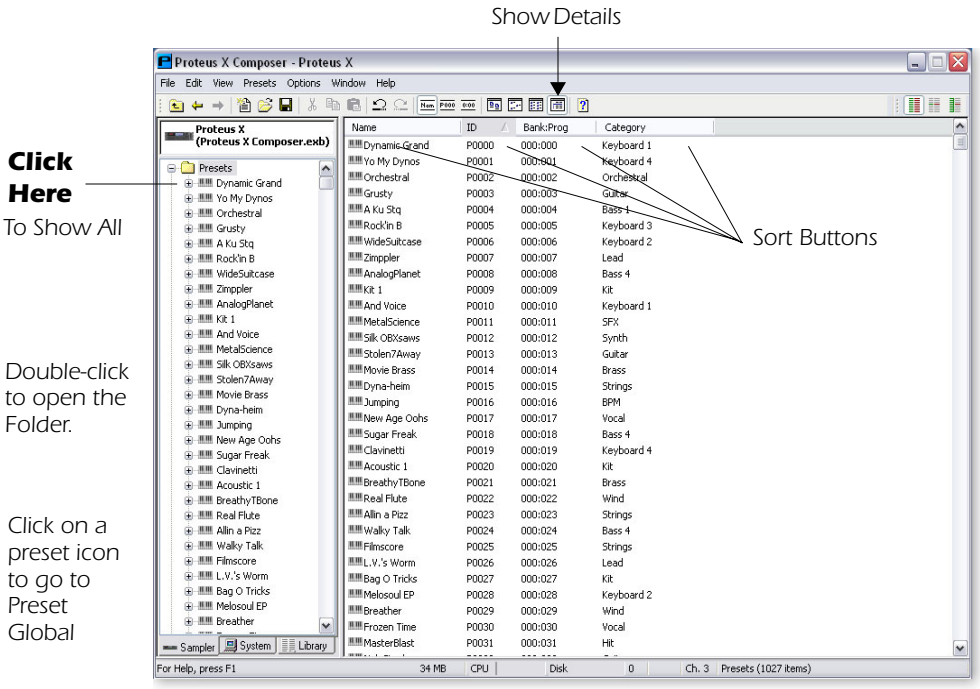
Note: Emulator 4 banks are accessed and loaded from the **System Tab**, rather than from the File menu.

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

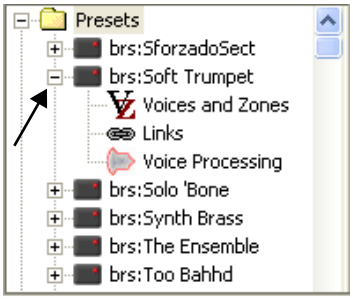
Opening a Preset

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.



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 and cords.

■ Please refer to chapter "2 - Proteus Architecture" on page 23 for complete explanations of the sub-modules.

5. Open the preset "**And Voice**" by clicking on the plus (+) sign next to the folder.
6. 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.

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

8. 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*

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

2. 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.
3. 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.
4. 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

5. Click on the **Val Win** tab. The following window appears:

6. This screen 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. **Ctrl-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.

Much More...

Now you've had a small taste of what you can do with the Voices and Zones section. Read chapter 5 to learn more about the Preset Editor. See page 55.

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.

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.

■ **Note:** See Groups on page 62 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.)

Modulating Cords

In the Proteus 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 (🡐).

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.

Proteus 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 **SamplePool** contains the raw sample data used. The **Bank** file contains all the voice, preset and multisample data that makes up the Proteus X bank.

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** 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 Proteus 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.

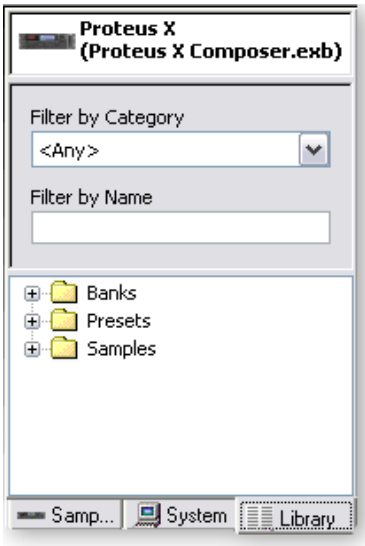
❖ **Cool Tip:** Categorize multiple Presets:
1. Go to the Tree
2. Click the Presets folder.
3. Select multiple presets.
4. Right-click and choose category.
5. Select category.

Find your Files

4. Select the Library tab at the bottom of the Tree. All Proteus X Banks, Presets, or Samples can now be accessed here. This can take a few seconds, especially with large banks.
5. Click on the **Presets** folder to view all the presets on your hard disks.
6. Select a Category of presets using the **Filter by Category** option.
7. Browse through the Bank, Preset, and Sample folders to find the sounds you want.
8. **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

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



Tip: *You can add Category tags to your own presets and samples by Right-clicking on them.*

4 - Multisetup / Global Controls

The Multisetup

The Multisetup is the top level of the Proteus 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.**

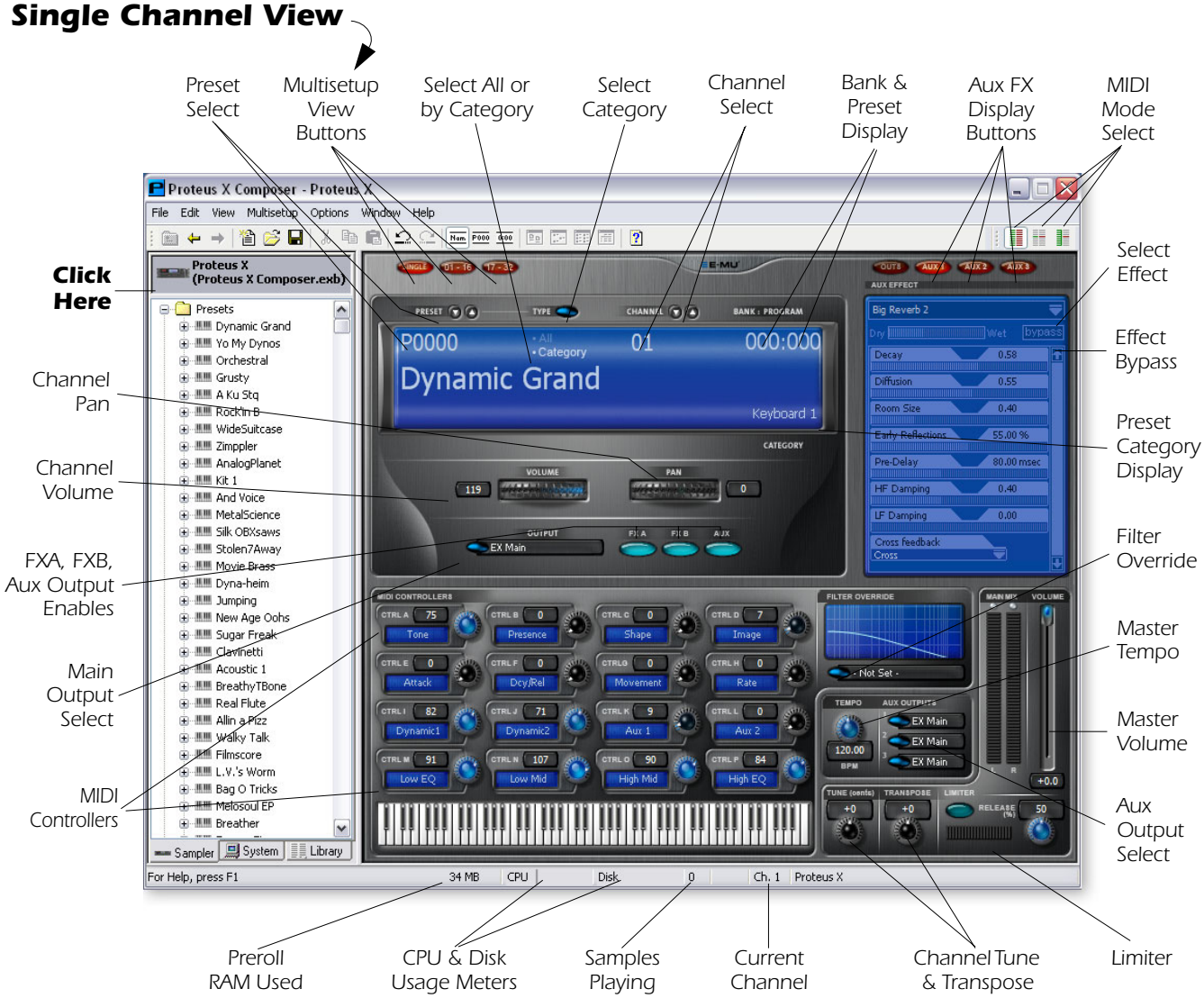
A multisetup assigns a preset, volume, pan position and output routing to each of the 32 MIDI channels. Multisetups can be saved and recalled for use with a particular song or sequence so that all the MIDI channels play the proper preset.

▼ 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 Proteus 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 three views of the Multisetup page. The **Multisetup View** buttons change the current view.

- **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.

Changing Presets (from Single View)

Single View is the default view when you first run Proteus 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

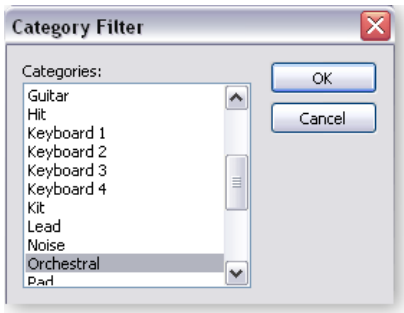
■ The Bank/Program Display shows the Bank and Program number you would use to select the preset from a sequencer.

▶ 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.



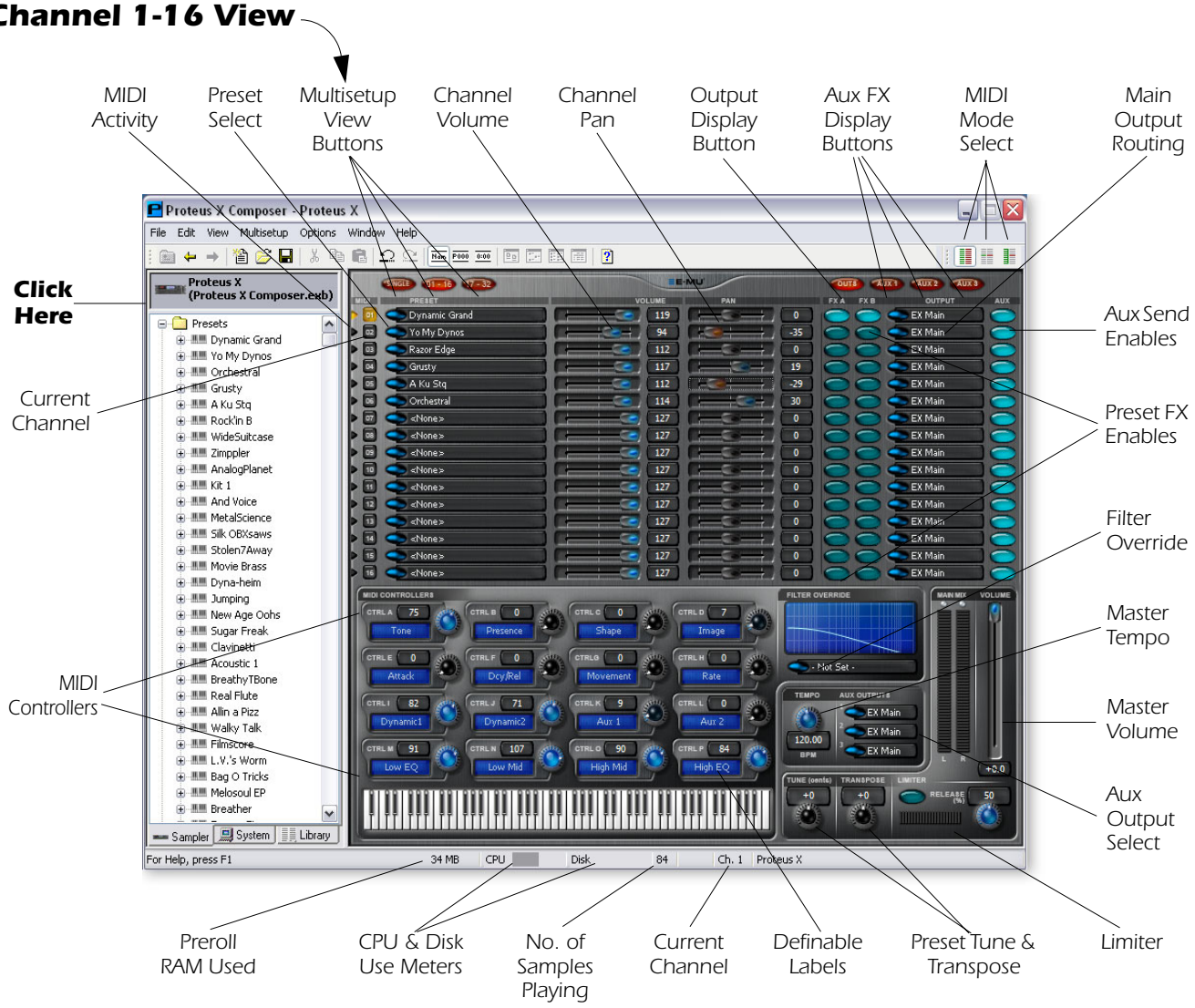
▶ Changing the Current MIDI Channel

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.

16 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.

Channel 1-16 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 p**ress 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 small keyboard at the bottom of the window 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 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.

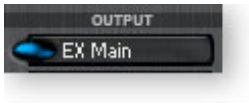
Channel Volume

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 the Proteus X.

▼ ***Important:*** *The act of creating PatchMix DSP ASIO mixer strips makes them available for Proteus 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

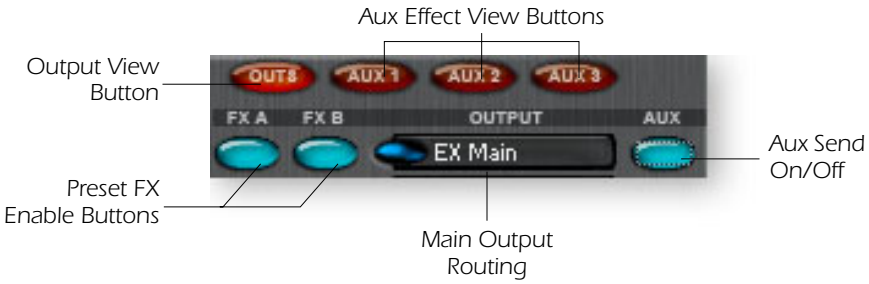
The buttons above the TV display select which of the three possible Aux Effect is currently displayed. Note: The Output View button is disabled in Single mode.

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 104 to see the exact location of this switch in the signal path.

■ The Output View button is only available when 1-16 or 17-32 is 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 "7 - Effects" on page 101 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 102 for complete details on how to add aux effects.

Global Controls

Master Volume Control & VU Meter

This is the master volume control for the Proteus 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 18.

The VU meter indicates the overall output level of the Proteus 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 the Proteus is chosen as the tempo source. Proteus 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 Proteus X tempo will lock to the MIDI clock tempo.

Master Tune

Master Tune adjusts the overall tuning of all presets so you can tune Proteus 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 the Proteus 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 semitones (1 octave).

Limiter

When playing the Proteus X you may sometimes hear digital clipping. This clipping occurs in the digital hardware and is not caused by the Proteus 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

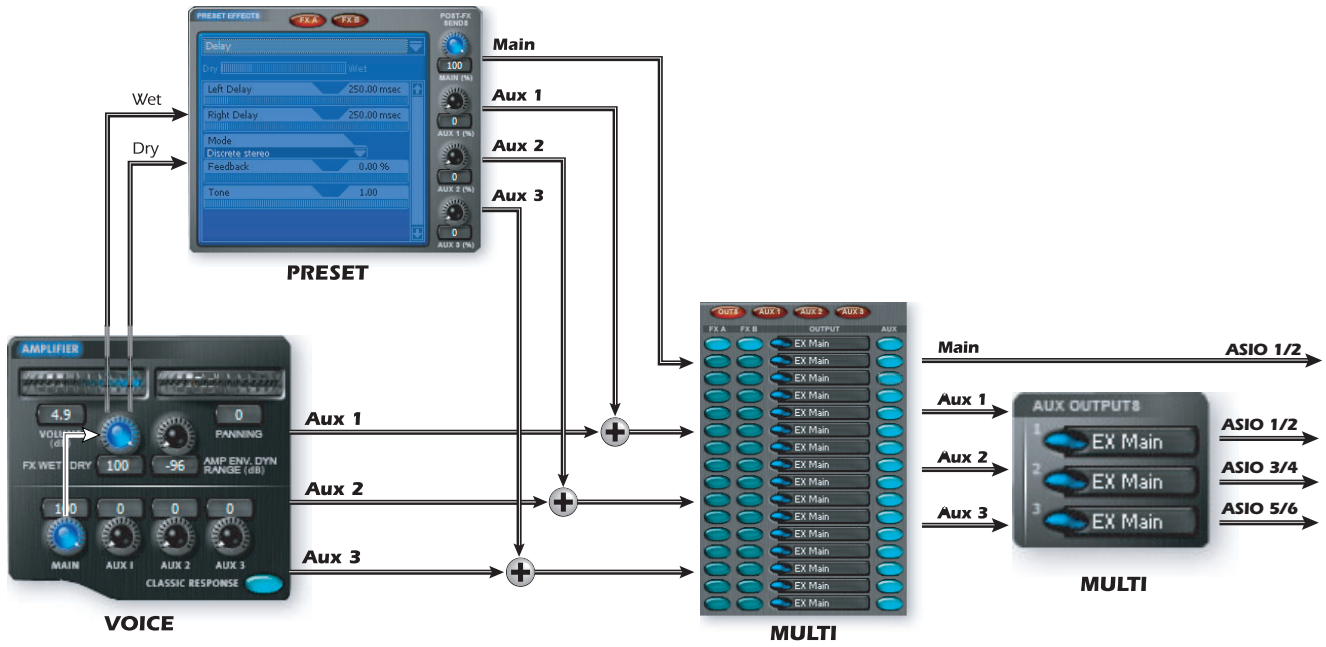
1. Press the Enable button, illuminating the button.
2. Set the Release knob to about 30% to start. Release controls how fast the limiter turns the volume back up after reducing it to avoid the clipping peaks.
3. Increase the release time if you hear obvious volume changes (pumping). Reduce the release 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 27) 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 "7 - Effects" on page 101 for complete descriptions of the effects routing.

Proteus X Aux Bus Routing



Proteus Sessions in PatchMix DSP

The Proteus X factory banks have been designed to be used with special Sessions in the E-MU Digital Audio System. These Sessions connect the Aux Outputs to PatchMix DSP mixer input strips containing PatchMix DSP hardware effects.

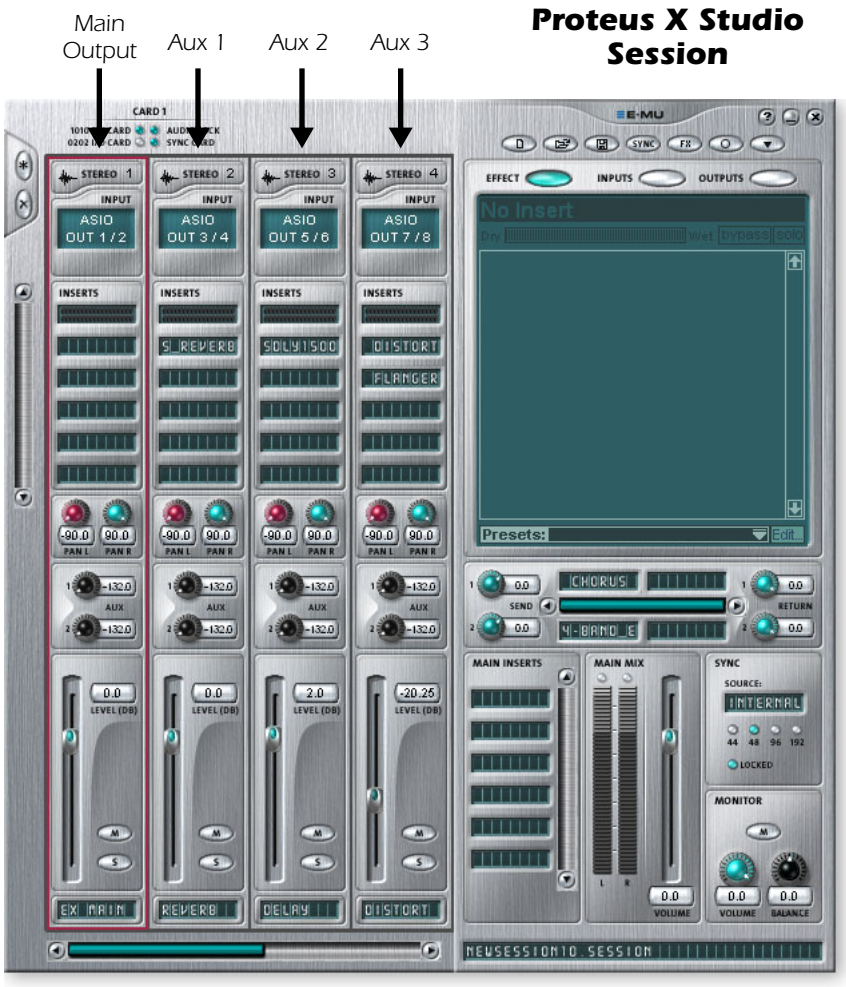
If a factory bank has the Aux Outputs set as shown in the diagram above, it is designed to use the PatchMix hardware effects. **If all the Aux Outputs are set to EX Main** it most likely uses the built-in **software effects**. You can easily add PatchMix DSP hardware effects to a bank that uses Proteus X software effects by simply switching an Aux bus to a PatchMix strip with effects. See "Block Diagram of the Effects Routing" on page 104.

IMPORTANT

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

▶ To Open the Proteus Session in PatchMix DSP

1. Open PatchMix DSP by clicking on the ☰ icon and select **New Session**.
2. Select the **Proteus X Studio** or **Emulator X Studio** session.
3. The PatchMix DSP screen below shows the Main and Aux Output connections from Proteus X.



The routing options in the Proteus 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 21) 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 56.

Mini Keyboard

The mini keyboard is used to audition presets and settings without changing the channel of your MIDI keyboard. The keyboard plays the currently selected MIDI channel at a velocity of 64. Select the current channel by clicking anywhere on the desired channel strip.

❖ Hold **Ctrl** and play the mini keyboard for a velocity of 10. Play the mini keyboard while holding **Shift** for a velocity of 127.

Preroll 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 15.

CPU Meter

This display shows how much of your computer CPU power is being used by the Proteus X.

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 17 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 Proteus X settings. Multisetups store the preset, volume, pan position, and output bus routings for 32 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 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 Multsetups

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 Proteus X standalone mode. If you wish to store extra multisetups you may do by using the Multisetup menu on the toolbar.

▶ To Save the Current Multisetup

1. Set up all the multisetup parameters the way you want them.
2. Select **Save** 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 save 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 **Proteus 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 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.

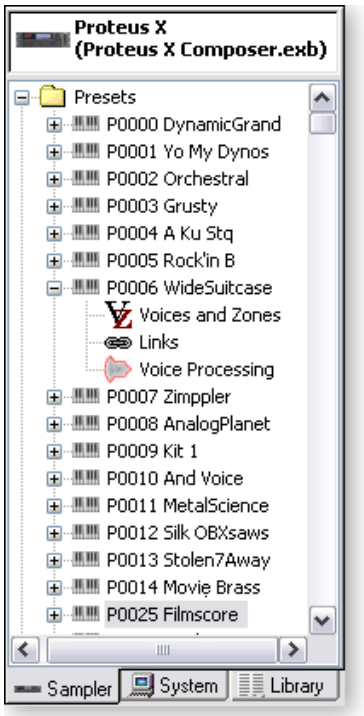
5 - Preset Editor

The Preset Editor allows 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 Proteus 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 Proteus 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.

Preset Globals

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

Transpose & Volume

Transpose works by shifting the keyboard position. The actual samples are not retuned. 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.

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

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

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 158 for additional information about Controllers A-P.

Tuning Tables

In addition to the standard equally divided octave tuning, Proteus 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.
Vallotti	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.
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
Werkmeister III	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.

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 terrible!

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**

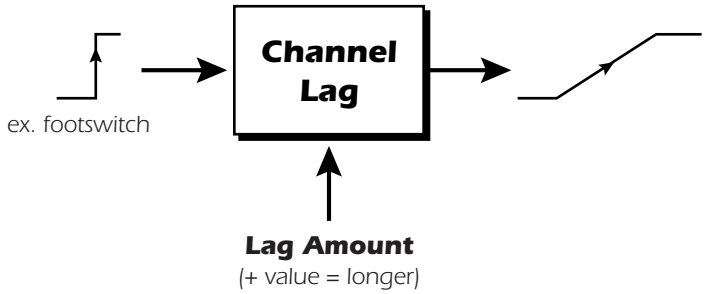
Preset Modulators
The Preset Modulators are special class of modulation sources in that they ***originate* at the *Preset Level*, but their output is used in the *Voice PatchCords*.** There are three types of preset modulation sources.

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. Used for "Hammond" style organ percussion.

Channel Lag 1 & 2 Rate
Like the Layer Lag processors *(described on page 98)* 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.

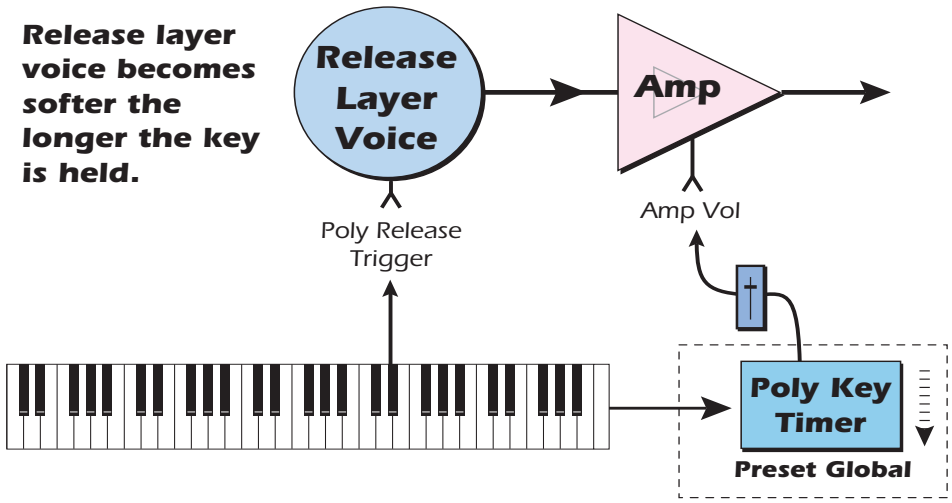
The Channel Lags could be used to slowly increase or decrease the speed of an LFO which in turn controls some other effect, perhaps left/right panning, pitch, or the filter. When a MIDI footswitch is used as the input, the lag acts to slow down the instantaneous change of the switch. The slowly changing output value can then be routed using a patchcord to crossfade between voices or 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 when the key is released. The value of this timer can be used on any voices assigned to the key after the key release. The Poly Key Timer is a special modulation source controlled by the key and not the voice.

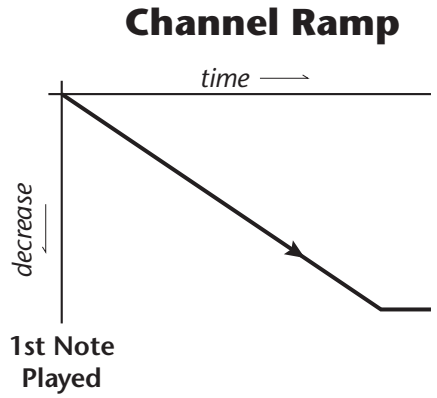
Poly Key Timer can be used to control the volume of a Release Layer in order to reduce its volume the longer a key is held. This control is commonly used on grand piano presets to lower the pedal damper volume the longer the keys are held.



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.



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 12 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 (F'switch 1 FF)	FX A Mod 1
Footswitch 2	FX A Mod 2
Flip-Flop Footswitch 2 (F'switch 2 FF)	FX A Mod 3
Thumbswitch	FX A Mod 4
Flip-Flop Thumbswitch (T'switch 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

Voices & Sample Zones

Mix/Tune Tab

Volume, pan, transpose, coarse tuning and fine tuning can be adjusted separately for each voice. See page 78 to understand the difference between transpose and coarse tune. You can also set the group number from this page.

The Mix/Tune parameters are the same controls displayed in the Voice Editor (page 78 & 90) and changes made to either page will appear in both places.

Voice Selection

A voice can be selected by clicking anywhere on the voice strip. Selection is shown by a red box around the voice strip.

1. **Select the first voice** in the "Voices & Zones" screen (red number).
2. Hold the **Shift** key to select contiguous voices. Hold the **Control** key to select non-contiguous voices. Press **Control-A** to select all voices.

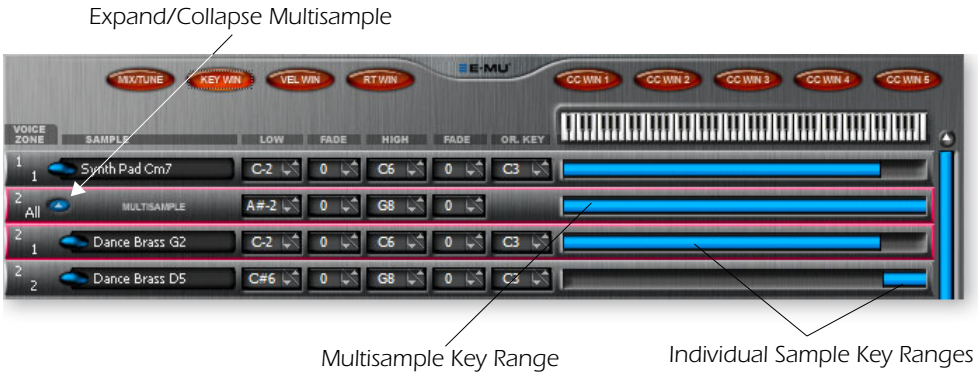
Note about multiple voice editing: If you open several voices for editing which have different voice parameter settings, the parameters in the different voices will remain as they were until you change them. That is, if you have three voices selected and you change the filter cutoff, all three voices 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 rearranged. If the voice contains more than one sample, the word, "Multisample" will 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 on the down arrow ▼ to display the individual samples and key ranges which comprise the multisample.
2. Click the arrow ▲ 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 Editor" window.
2. Whenever you select that group number in the Voice Editor screen, 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.

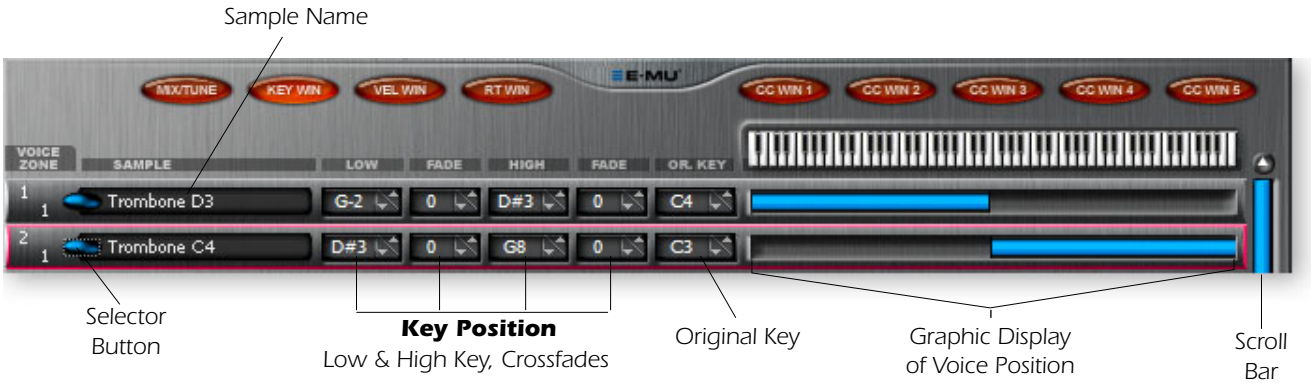
▶ To Select All Voices for Editing

- From the "Voices & Sample Zones" window, press Control-A.
- From the "Synth Editor" 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.

Key Window Page



The Key Range Window allows you to edit the keyboard range assignment 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 the correct location. See page 190.

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:

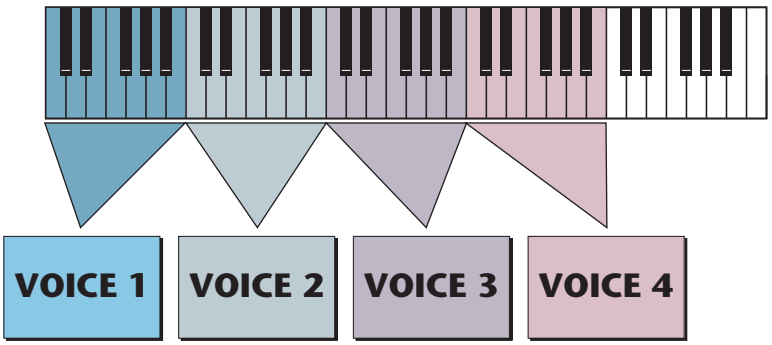
- **To Set the Length of the Bar:** Left-click and drag from either endpoint.
- **To Create a Fade:** Left-click and drag inward while holding the **Ctrl** key.

The voice will gradually fade-out in volume below F2 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.

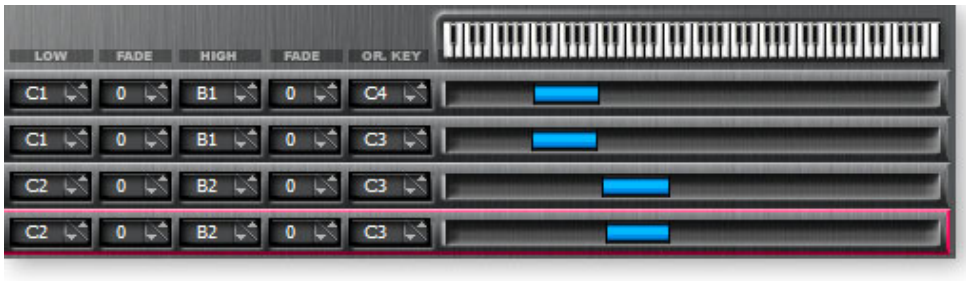
Another way of visualizing this assignment is shown below.



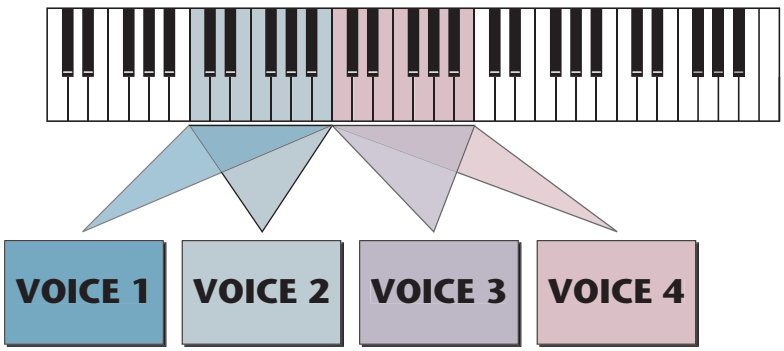
❖ Samples can be arranged in a multisample voice in exactly the same way that voices are arranged on the keyboard. They can be arranged side by side on the keyboard or stacked on top of each other to create layered sounds, all within a single voice.

▶ 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.

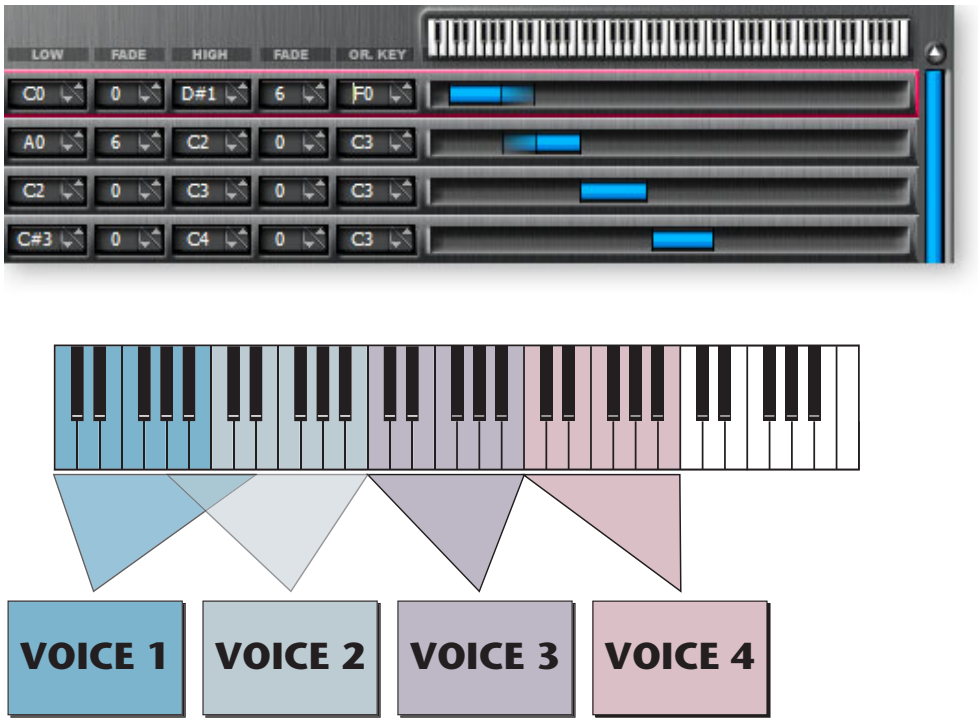


▶ To Positionally Crossfade

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.

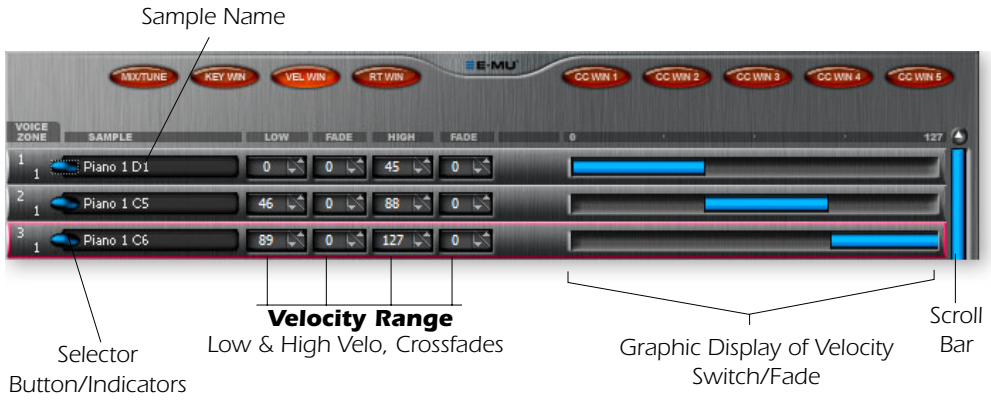
As the voice boundaries are crossed playing up the keyboard, the lower voice gradually fades out as the upper voice fades in.

■ 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.



Velocity Window

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 example screen on the previous page, 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.)

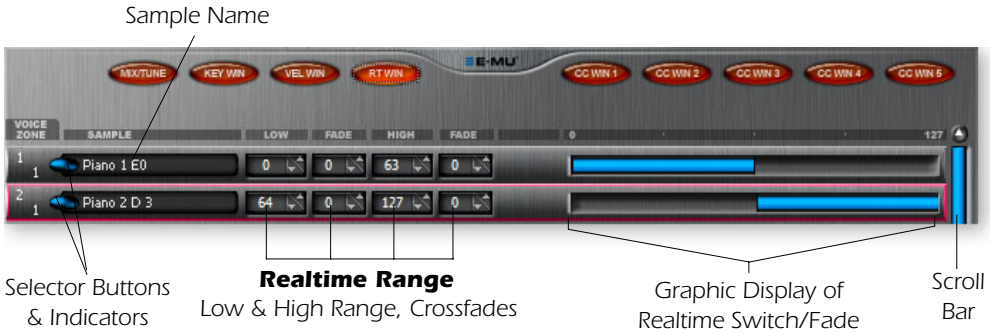
▶ To Velocity Crossfade Voices:

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 faces out the other fades in.
4. Play the keyboard and fine tune the crossfade points for the desired response.

Realtime Window

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.



▶ 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 faces 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 **Crtl** 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** (Crtl+Enter) from the Preset menu. The Voice Editor screen 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** (Crtl+Enter) from the Preset menu. The Voice Editor screen 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.
8. Select **Amp Crossfade** as the destination. Set the **Cord Amount** to +127.
9. 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.

Continuous Controller Window Buttons

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 to the right of the Assignment screen. 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.

▶ 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.

Voice & Sample Zone Utilities

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

Voice Utilities

Open Voices

This function opens the voice editor for the currently selected voice(s).

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

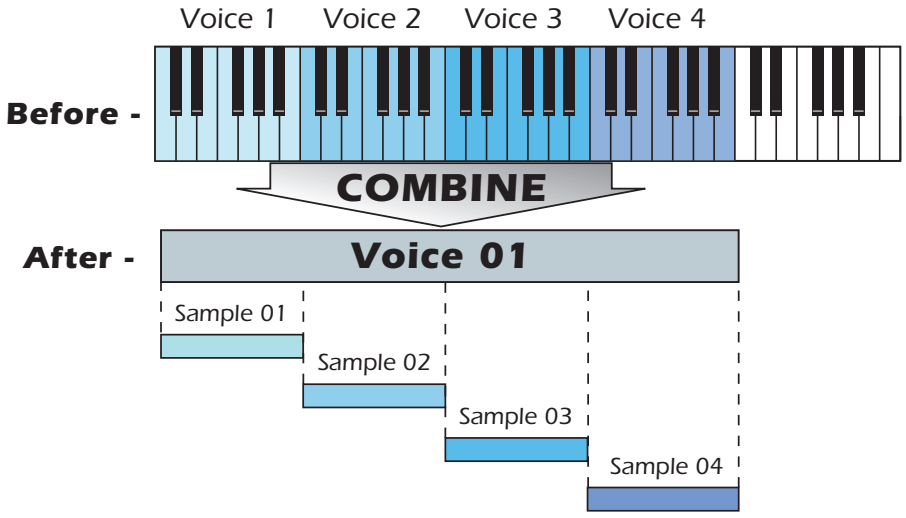
This function deletes the selected voice, but does not erase the samples.

Duplicate Voices

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

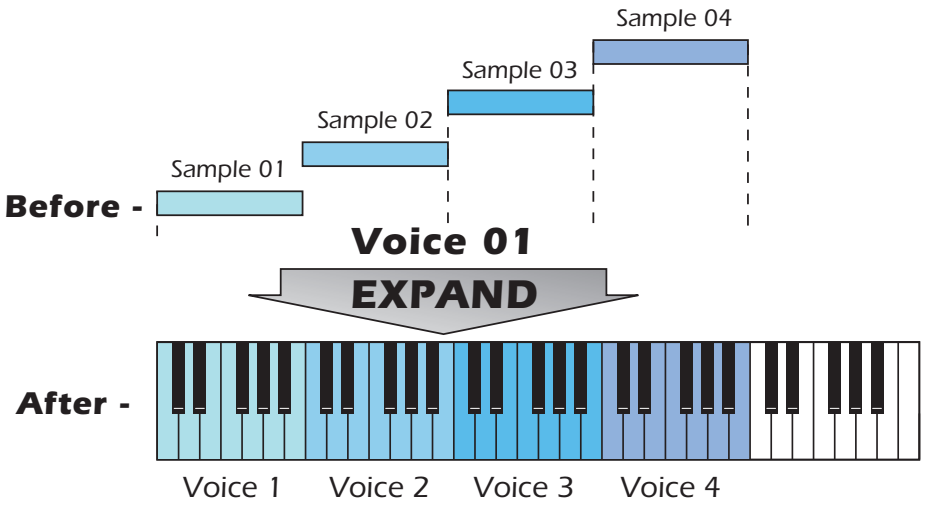
Combine Voices

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.



Sample Zone Utilities

A Sample Zone is the keyboard mapping of the samples used in a voice. Use these utilities when editing multisample voices.

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 multisample.

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.

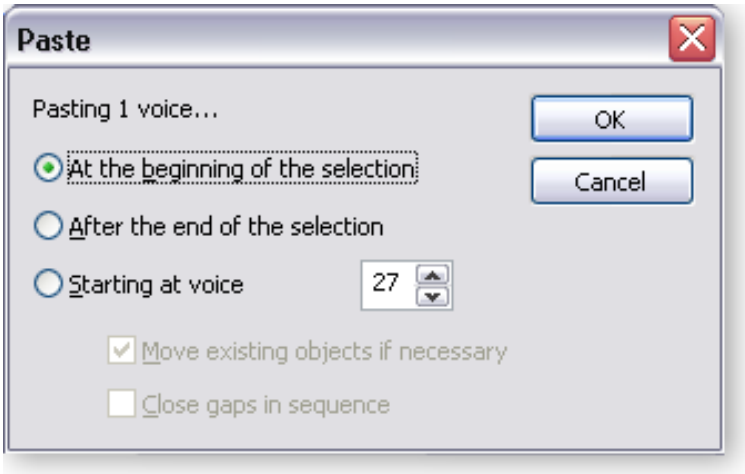
Edit Menu Utilities - Cut, Copy & Paste

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)

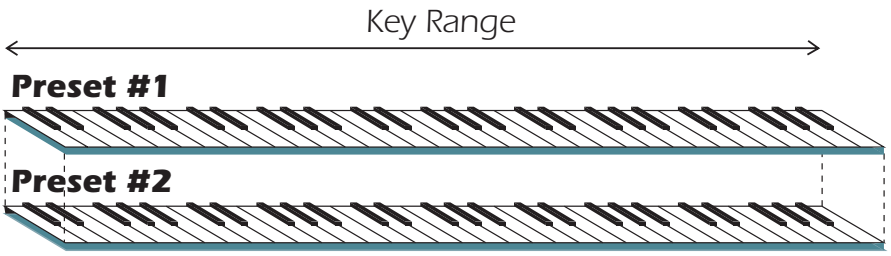
1. **Select the voice(s)** you wish to copy. (Hold Shift to select contiguous voices.)
2. Select **Copy** from the Edit menu.
3. **Open a new preset** if you want to paste between voices.
4. Select the **Voices and Samples** page.
5. Select **Paste** from the Edit menu. The following dialog box appears.



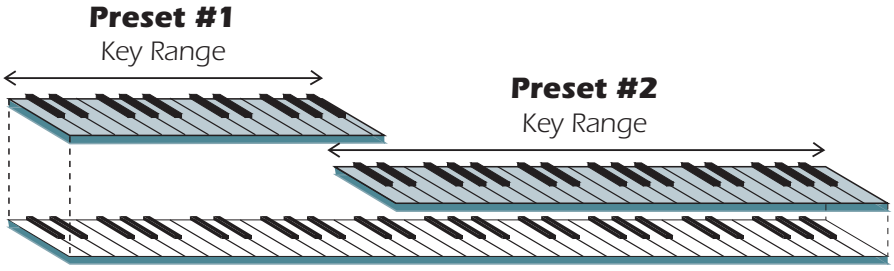
6. You have the option of pasting the voice(s) before the currently selected voice, after the currently selected voice or starting at a specified voice number. This feature allows you to rearrange your voices any way you want.
7. Press **OK** to paste the voice or **Cancel** to cancel the operation.

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.



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.

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

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.

Link Utilities

These three controls are accessed from the main menu.

New	Creates a new link.
Delete	Deletes the selected link(s).
Duplicate	Duplicates the selected 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.

Mix/Tune Window

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

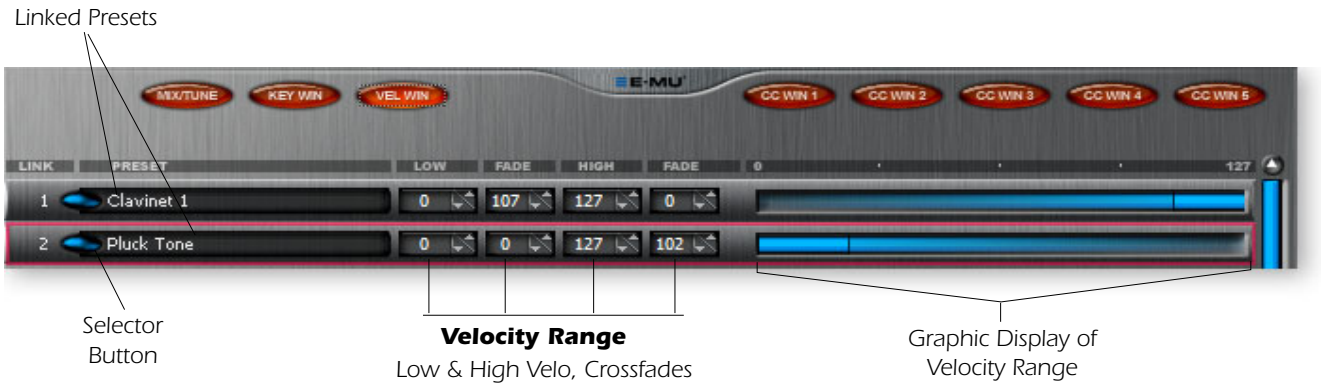
- **Volume** — Adjusts the volume of the linked preset.
- **Pan** — Sets the stereo pan position of the linked preset.
- **Transpose** — Transposes the linked preset by shifting the keyboard position. (See page 78.)
- **Fine Tune** — Allows you to detune the linked preset by ±100 cents. (1 semitone)

Key Window

The key window works just like the key window in the Voices and Zones section (page 61) and allows you to set the keyboard range for each linked preset.

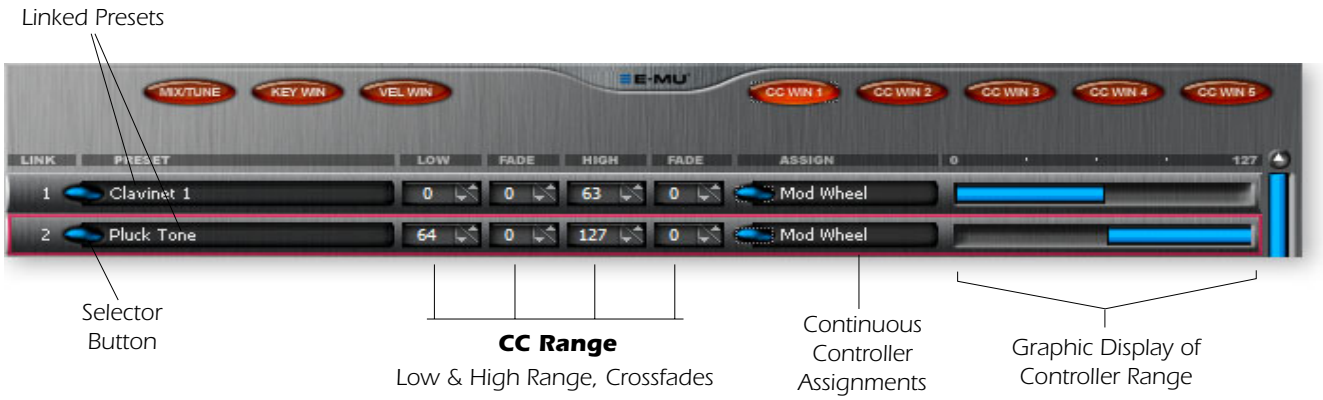
Velocity Window

The velocity window works just like the velocity window in the Voices and Zones section (page 61) 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 68) 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 Editor / Synthesizer Level

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:**

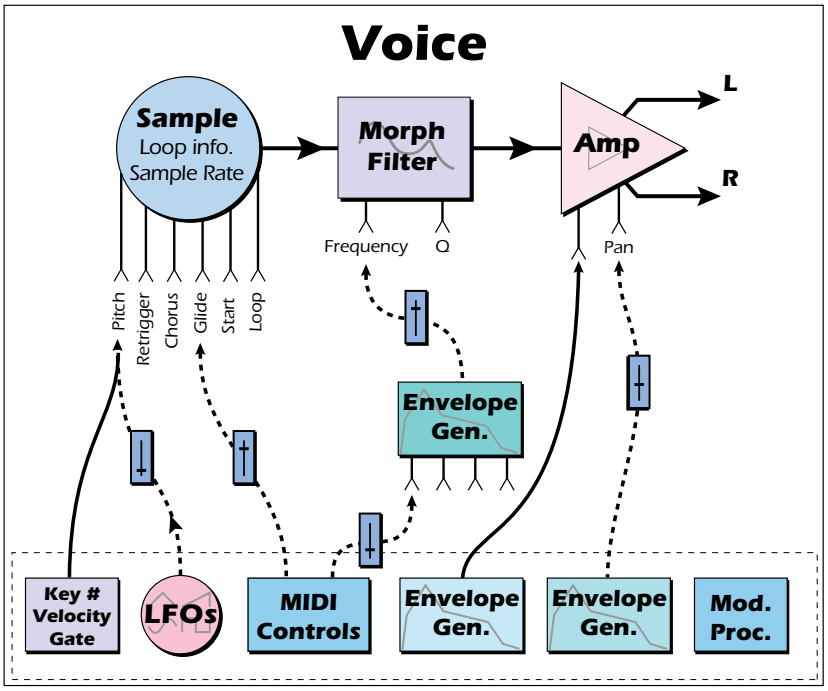
- Assign voices to Groups in the Voices and Zones page, then select the group.
- Select the desired voices from the Voices and Zones page.
- Drag the **range bar** below the little keyboard to select a range of voices.
- Hold **Crtl+Alt** and play your MIDI keyboard to select voices (with IntelliEdit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

❖ See Groups on page 62.

❖ IntelliEdit is enabled in the MIDI Preferences dialog box. See page 19.

Simplified Proteus X Signal Path

A single Proteus 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 the Proteus X and make programming much easier.

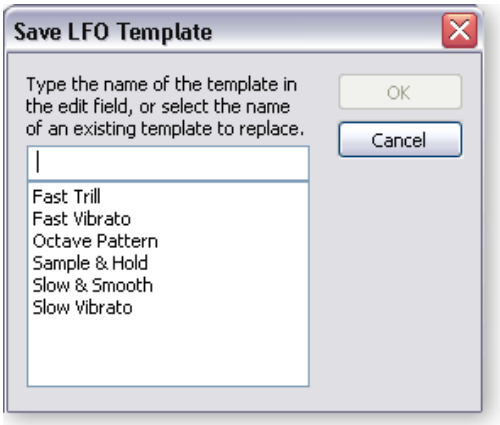


Overview of the Voice Editor

The voice editor 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 editor or 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 the Proteus X application so they're always available for use. Here's how they work:



▶ To Create a Template

1. Set the controls as desired for a particular module.
2. **Right-click** with the mouse anywhere inside the module border. 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 with the editor program, ready to use when you open another bank.

▶ To Recall a Template

1. **Right-click** with the mouse anywhere inside the module border. 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 following popup dialog box appears.

2. **Select the Template** you want and click **OK**.

▶ 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.

▶ To Change the Order of the Templates

1. **Select the desired template** from the Organize Templates dialog box.
2. Click **Move Up** to move the Template one space up in the list.
3. Click **Move Down** to move the Template one space down in the list.
4. Click **OK** to close the dialog box.

▶ To Save All the Voice Parameters as a Template

1. Select **Voice Processing Templates** from the Menu Bar.
2. Choose **Save Voice Processing Template...** from the menu. A pop-up dialog appears asking you to name your new Template.
3. Name the template.
4. Press **OK** to save the Template or **Cancel** to cancel the operation.

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.

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 semitones. 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 61) 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 semitones. 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 61) and changes made to either page will appear in both places.

Fine Tuning

This function allows you to change the tuning of the voice in 1/100 cent intervals (100 cents = 1 semitone). The fine tuning range is ± 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 61) 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.

❖ 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.

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 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

▪ Set chorus amount to 1% and use chorus width as a pan control which does not span the entire L/R spread.
▪ 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.
▪ To avoid L/R pitch drift, use "~" type of modulation in the cords selection box.
▪ Modulate the ITD to create "Hass-Effect" panning, (i.e. apparent location shift using slight delays in the L-R channels.)
▪ 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. A glide curve setting of 0 equals linear. Higher values make the curve more exponential. 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.

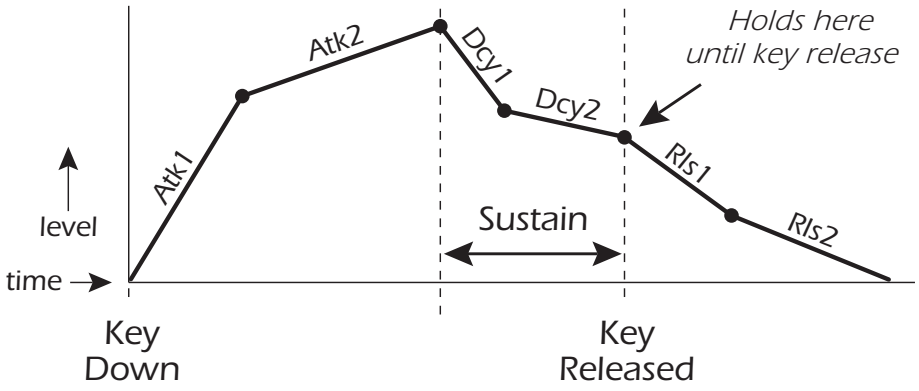
Sample Offset

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. See "Pre-roll" on page 16.

Amplifier, Filter & Auxiliary Envelopes

Proteus X contains three envelope generators per voice. The Amplifier Envelope is "hard-wired" to control the DCA (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.

Time-based and BPM Envelopes

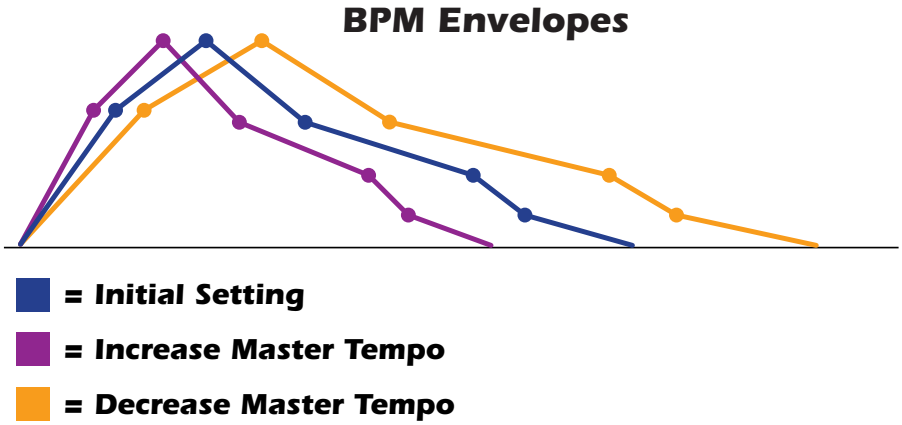
Envelope times can be time-based or BPM-based. Time-based envelopes have an absolute time setting, such as 1 second. BPM-based envelope times are divided down from the Master Tempo setting on the Multisetup page. The Master Tempo scales the time of the BPM-based envelope segments.

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

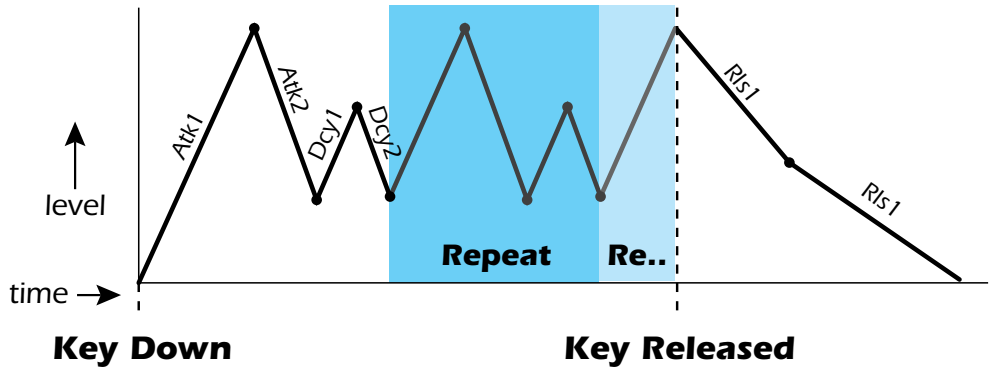
BPM envelopes are controlled by the Master Tempo *(located in the Multisetup window)*. The Master Tempo rate scales the time of the BPM envelope segments. The Master Tempo can also be derived from an external MIDI clock to synchronize the envelope times with external sequencer or arpeggiator tempo changes.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



Envelope Repeat

The Envelope Generators can also 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.

■ Only the Filter and Auxiliary Envelopes have the repeating feature.

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

Filter Controls

You can choose from 53 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.
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.

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
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->1octave	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 87.*
Dual EQ + Lowpass Morph	06	PROG	*Please refer to the detailed description on page 89.*
Dual EQ Morph + Expression	06	PROG	*Please refer to the detailed description on page 88.*
Peak/Shelf Morph	06	PROG	*Please refer to the detailed description on page 89.*
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 “Ringy” Q response. Many Q variations.

Filter Name	Order	Type	Description
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. I094, Q =60
DreamWeava	12	FLG	Directional Flanger. Poles shift down at low Q and up at high Q.
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

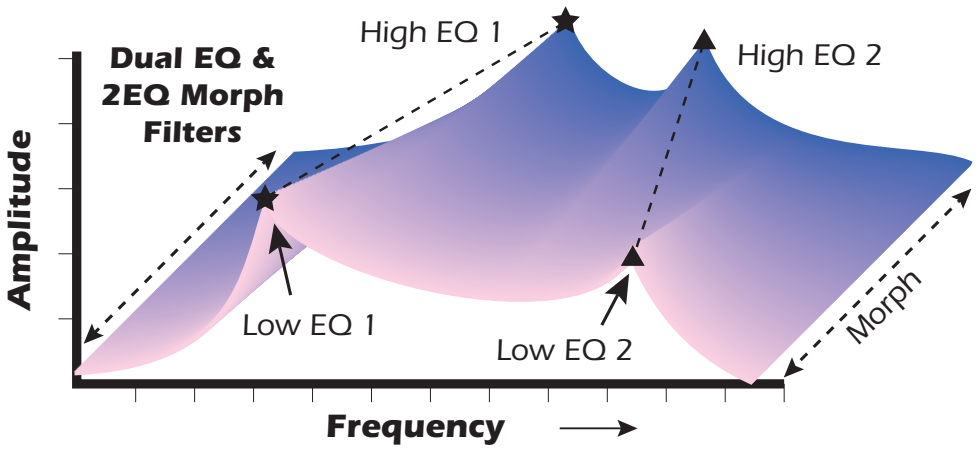
There are four user-programmable morphing filters which have the ability to change their response over time. Now you can design your own custom filters that morph between filter types. Take the time to learn about these powerful filters because they are extremely useful and sound great.

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.

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.)

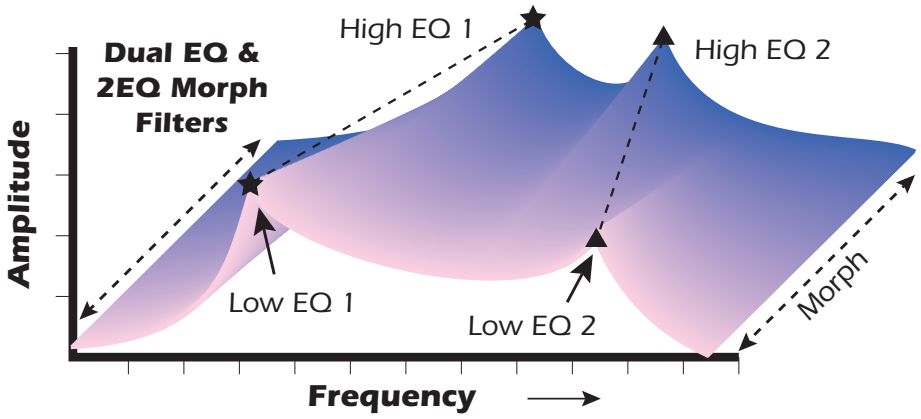


As the Morph value is increased, the filter interpolates from the Low to High Frequency settings.

2EQ Morph + Expression

- Morph = Filt. Freq.
  Express'n = Filt. Res.

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.



2EQ + Lowpass Morph

- FC/Morph = Filt. Freq.
  LPF Q = Filt. Res.

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.

Peak/Shelf Morph

- Morph = Filt. Freq.
  Peak = Filt. Resonance

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.

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).

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 61).

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 61).

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.

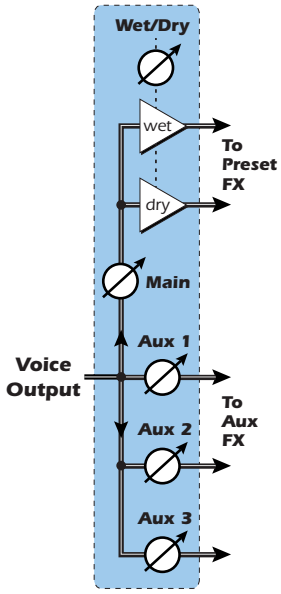
Response

This control changes the amplitude curve of the voice. Classic response emulates the volume curves of classic E-MU synthesizers.

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.

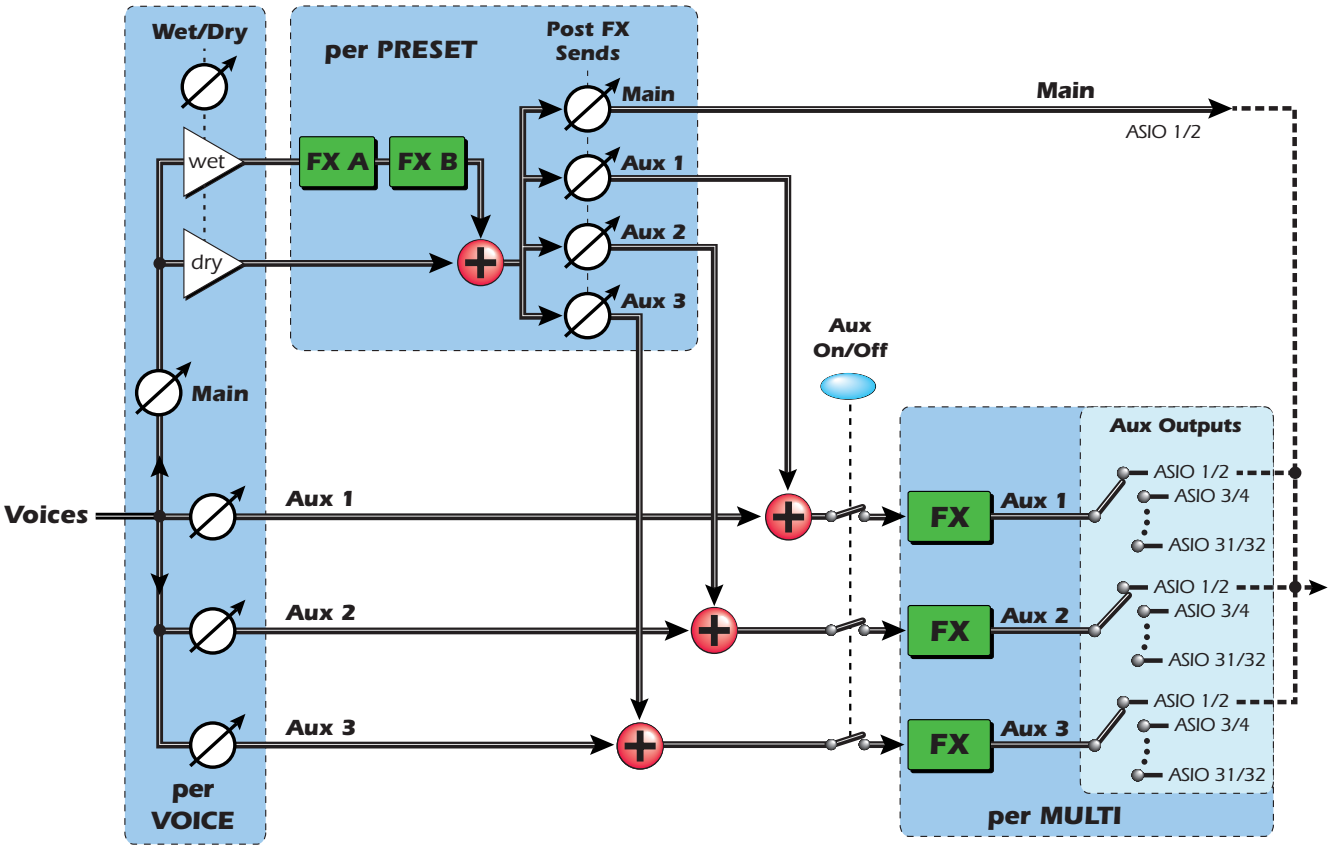


Voice Effect Sends

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 55. 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.

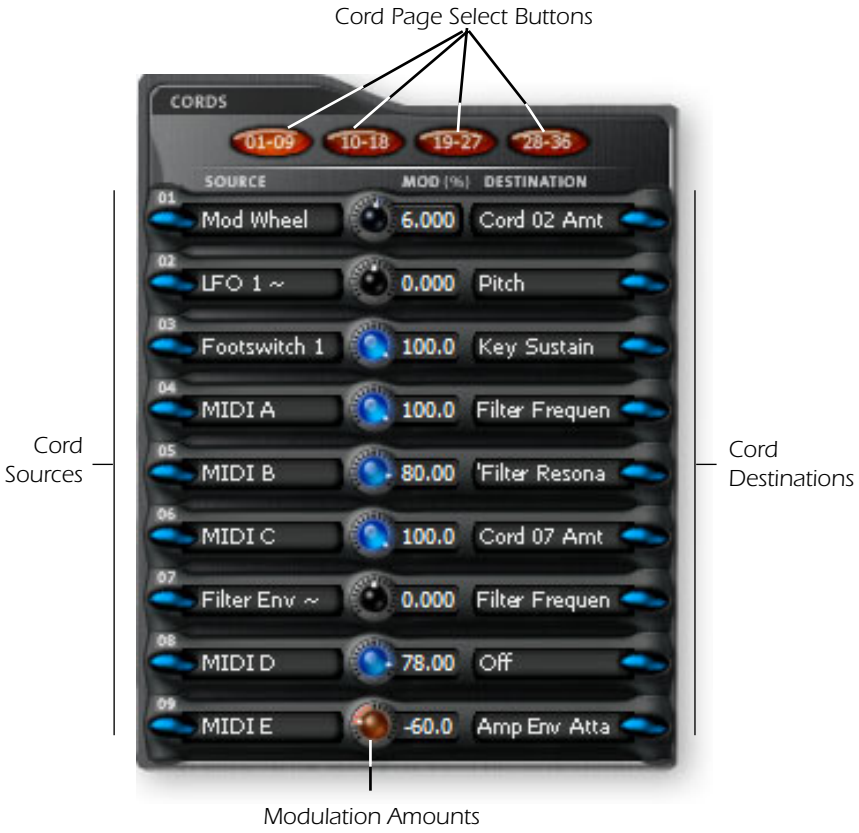


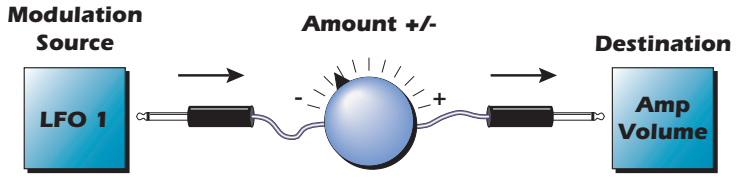
The Voice Effect Send levels and the Wet/Dry mix can be controlled using the voice modulation cords. See "Send Aux 1-3" on page 94.

See also "Controlling the Preset Effect Amount from the Voice" on page 109.

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
Gate	This value goes to full value as long as a key is depressed.
Pitch Wheel	MIDI data from your keyboard's pitch bender.

Modulation Sources

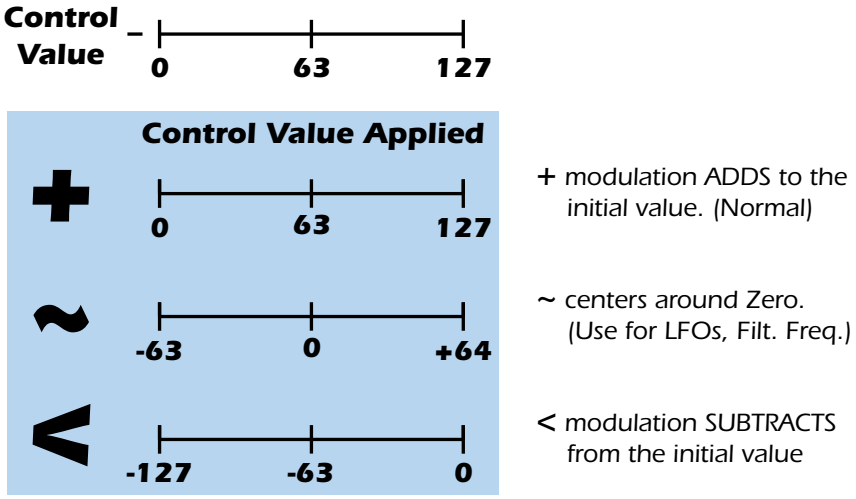
Source	Description
Mod Wheel	MIDI data from your keyboard's modulation wheel or lever.
Pressure	Aftertouch MIDI data (channel pressure)
Pedal	Footpedal MIDI data (normally controller #4)
MIDI A - P	16 assignable MIDI continuous controllers See page 21.
Footswitch 1 & 2	MIDI footswitch data (normally controllers 64 & 65)
F-F Footswitch 1&2	MIDI footswitch inputs which latch on or off with each press.
MIDI Volume	MIDI Continuous Controller data on channel #7
MIDI Pan	MIDI Continuous Controller data on channel #10
Key Glide	A smoothly changing control source based on the Glide Rate and the interval between the last two notes played.
Amp Envelope	Amplifier Envelope Generator (+, ~, -) See page 95.
Filter Envelope	Filter Envelope Generator (+, ~, -) See page 95.
Aux Envelope	Auxiliary Envelope Generator (+, ~, -) See page 95.
LFO 1 & 2 (+, ~)	Low Frequency Oscillators 1 & 2
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 98.
Lag Processor 1&2	Lag processors slow down rapid changes. See page 98.
Channel Lags 1&2	Lag processors that operate at the channel level. See page 58.
Channel Ramp	Slope generator that is triggered by the first key pressed. See page 60.
Poly Key Timer	Outputs a value based on the time a key is held down. See page 59.
Clock Divisors	MIDI clock is divided to the specified value and used as a modulation source. See page 148.
DC Offset	A fixed value that can be scaled by the Cord to offset a value.
Summing Amp	The Summing Amp is useful for mixing several modulation signals so they can be controlled from a single Cord amount. See page 151.
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 151.
Absolute Value	This processor inverts negative values and leaves positive values untouched so that only positive values are output. See page 151.
Diode	This processor allows only positive values to pass. Negative values are blocked. See page 151.
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 151.
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. See page 151. *(Hint: Gain 4x can increase number of steps or step size.)*
Gain 4x	This processor amplifies the modulation input by a factor of 4.

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 79.
Sample Start	Controls the sample start offset point. See page 80.
Sample Loop	Controls the sample loop length. See page 98.
Sample Retrigger	A negative-going, zero-crossing value retriggers the sample from the sample start.
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 66.
Send Main	Controls the level of the voice sent to the Preset FX and the Main Output. See page 104.
Send Aux 1-3	Controls the level of the voice sent to the Aux Send 1, 2 or 3 FX processors. See page 104.
Wet/Dry Mix	Controls the Wet/Dry mix sent to the Preset FX. See page 104.
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.
Lag Processor (0&1)	Inputs for Lag Processors 0 & 1. See page 98.
Summing Amp	Input to the Summing Amp. See page 151.
Switch	Input to the Switch processor. See page 151.
Absolute Value	Input to the Absolute Value processor. See page 151.
Diode	Input to the Diode processor. See page 151.
Flip-Flop	Input to the Flip-Flop processor. See page 151.
Quantizer	Input to the Quantizer. See page 151.
Gain 4x	Input to the 4x Amplifier. See page 151.
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.



+ modulation ADDS to the initial value. (Normal)

~ centers around Zero. (Use for LFOs, Filt. Freq.)

< modulation SUBTRACTS from the initial value

Summing Nodes

All the modulation inputs on Proteus 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

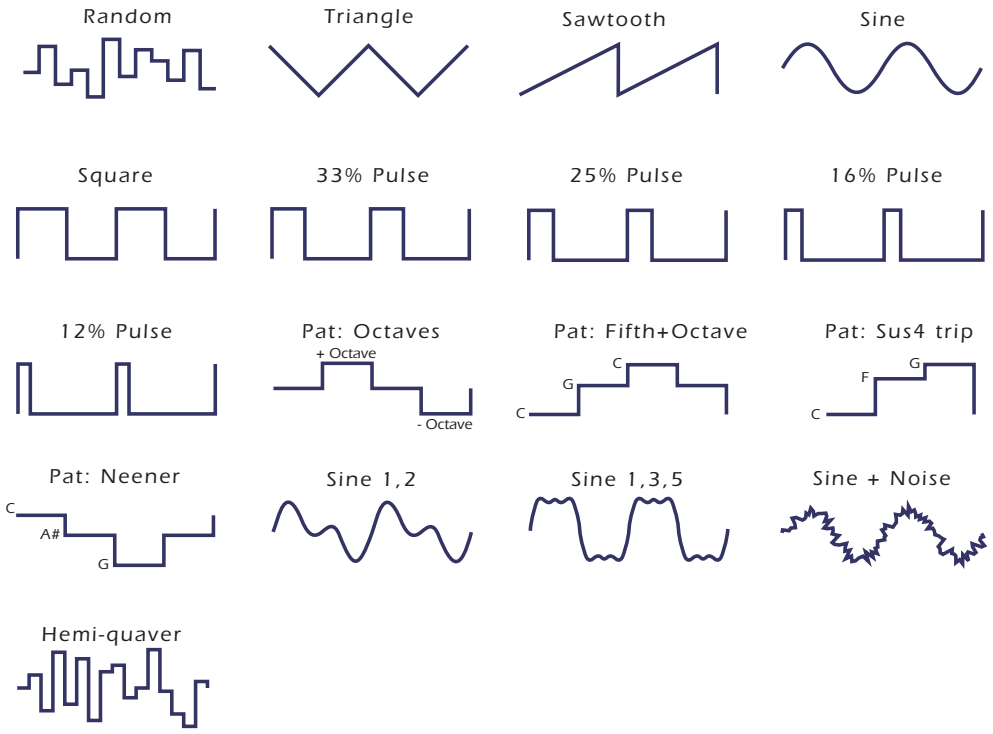
- Real-time events happen first on lower numbered Cords, so it is usually a good idea to put important functions on lower numbered cords and to place components of complex modulations patches in the proper order in the Cords section.
- Negative Cord amounts invert envelopes and LFO waveforms.
- To invert the effect of a controller knob, set its Cord to -100 and add DC with a Cord value of +100.

LFO 1 & 2

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. Proteus X has two multi-wave LFOs for each voice, with 17 different waveforms which are diagrammed below.

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.

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



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 an LFO 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 rate 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

LFO Tricks & Tips:

The Random LFO wave is truly random and is different for each voice and layer.

The Pattern (Pat) waveforms will sound the same on different layers and voices.

When the PatchCord amount for an LFO is a negative value, the LFO shape will be inverted. *(A rising sawtooth turns into falling sawtooth.)*

Sine + Noise is very useful for simulating trumpet and flute vibrato.

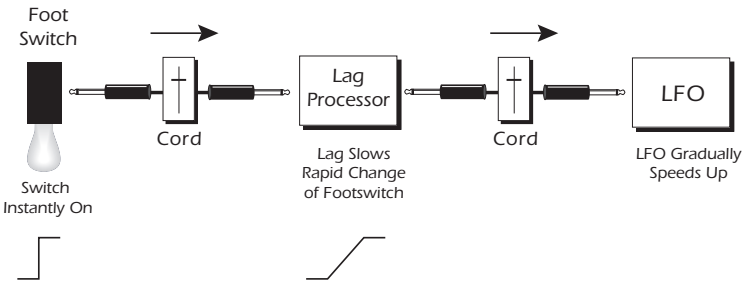
When routing Hemi-quaver to Pitch...

+38 = major scale -38 = phrygian scale
+76 = whole tone scale (+38) + (+76) = diminished scale (two cords)
odd amount = S+H sound.

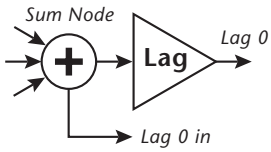
Try combining the Pattern LFOs, or controlling the amount of one with another, or combining them with the clock divisors.

Lag Processors 1 & 2

Lag processors can be inserted in a modulation patch to inhibit rapid changes in the modulation source or cause it to lag behind the input. The amount (rate) of lag is adjustable from 0 to 10. A setting of 0 will cause the least amount of lag. 10 is the slowest lag.



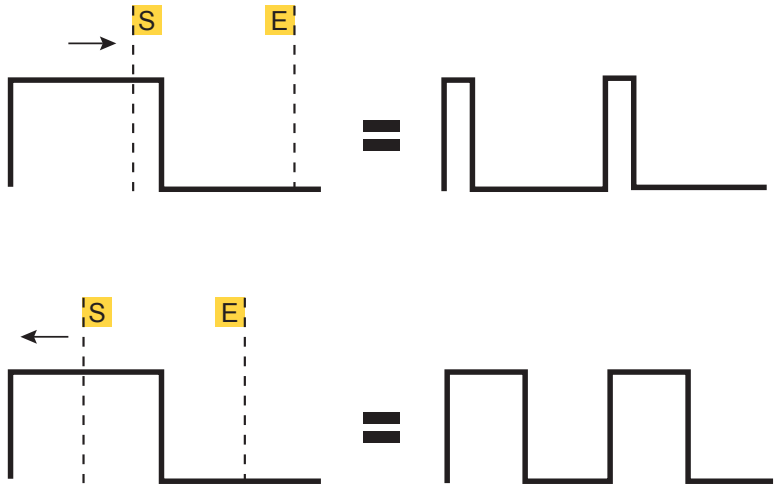
Lag processors are used to slow down and smooth rapid changes in control signals. In the example above, the instantaneous switch change becomes a smooth transition.



The Lag Processors can also be used as 2 additional summing amps. Summing amps are used to add several modulation sources together so the output can be controlled by a single Cord Amount. The output of Lag0in and Lag1 modulation sources are equal to the sum of any cords connected to the Lag in destinations. The summing node is located before the lag processor as shown in the diagram.

Sample Loop Modulation

Although there is no initial control, the sample loop position can be modulated using a Cord. See "Modulation Destinations" on page 94. The loop length remains constant as the loop is modulated. This feature can be used to implement pulse width modulation, and other wavetable scanning synthesis techniques.



By modulating the loop over a single cycle of a square wave, it can be "pulse width modulated." Other more complex waveshapes can also be modulated in this way.

Assign Group

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.

Poly All	Notes are played polyphonically with dynamic channel assignment, using all 128 channels.
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.

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 only triggers 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 16) 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. Retriggers 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.
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. 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.

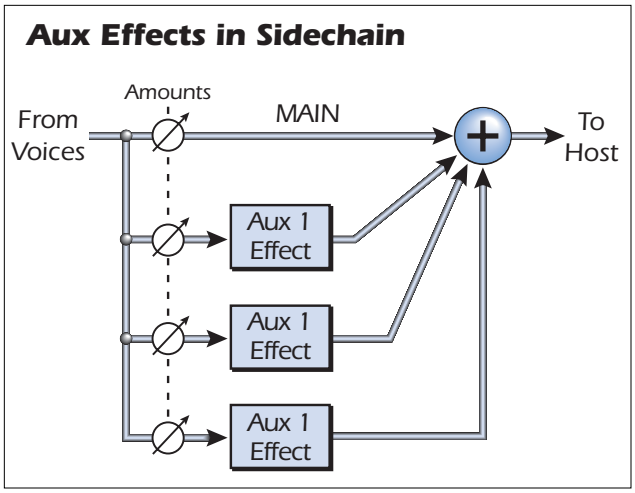
7 - Effects

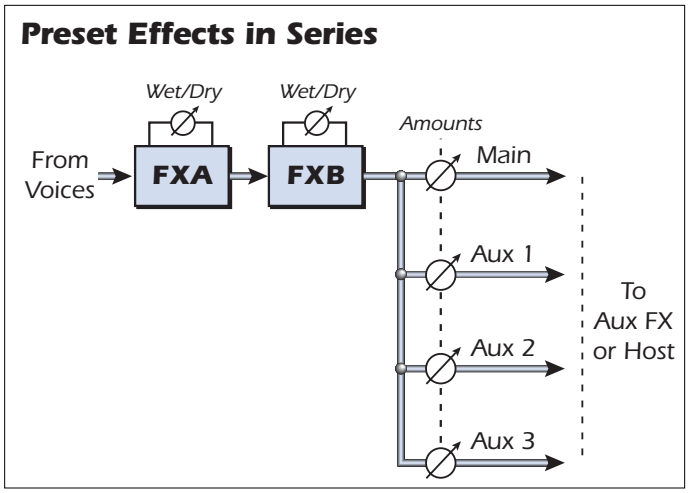
Overview

In addition to the hardware-based effects of PatchMix DSP, Proteus 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 Proteus 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 32 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 21 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 Proteus X soft effects. **These instructions explain how to add Aux Effects to a Proteus X bank which does not already have soft effects programmed.**

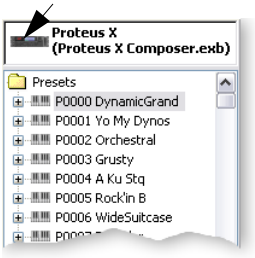
The Aux Effects are designed to be used as sidechain effects applied on all presets in the bank. See the diagram on page 104 for a complete description of the effects routing.

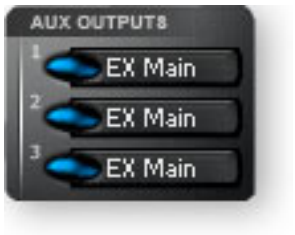
❖ PatchMix DSP hardware effects routings are already programmed into most Proteus X banks

▶ How to Add an Aux Effect

1. Select the **Proteus X Studio** session in PatchMix DSP.
2. Close PatchMix DSP
3. Select the **Multisetup Page** by clicking the **Proteus 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**. (This bypasses the PatchMix DSP hardware effects and sets up the Aux Outs to be used as sidechain effects.)

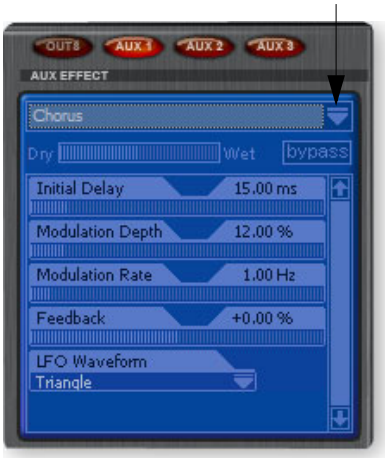
Proteus X icon





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.

Effect Select icon



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.

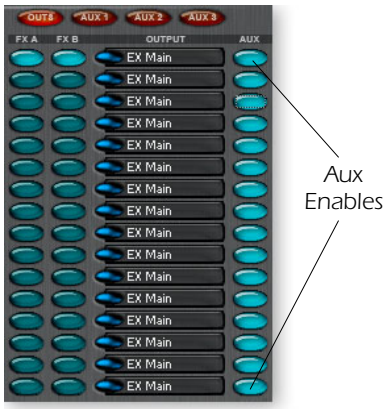
■ When creating your own presets and banks, we recommend you follow the FX routing model on page 108.

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 M**ultisetup Page** by clicking the Proteus 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.



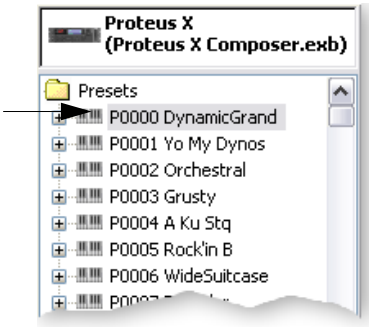
■ The Aux buttons turn all three Aux busses On or Off.

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.

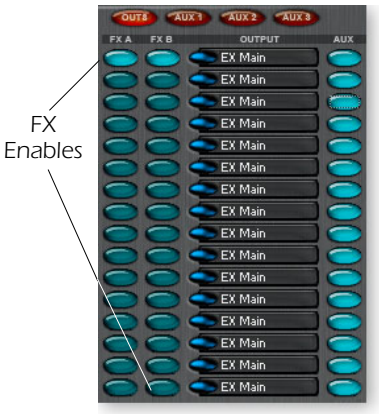


4. **Play the keyboard.** You should now hear the selected effect as part of your sound. If you don't hear effects, make sure the FXA & FXB enable buttons are on for the MIDI channel you are using. See page 104.
5. Adjust the **Wet/Dry mix** to control the effect amount.
6. Adjust the effect parameters until you find the sound you want or select a Template. See page 76.
7. **Save the bank** if you want to keep your changes.

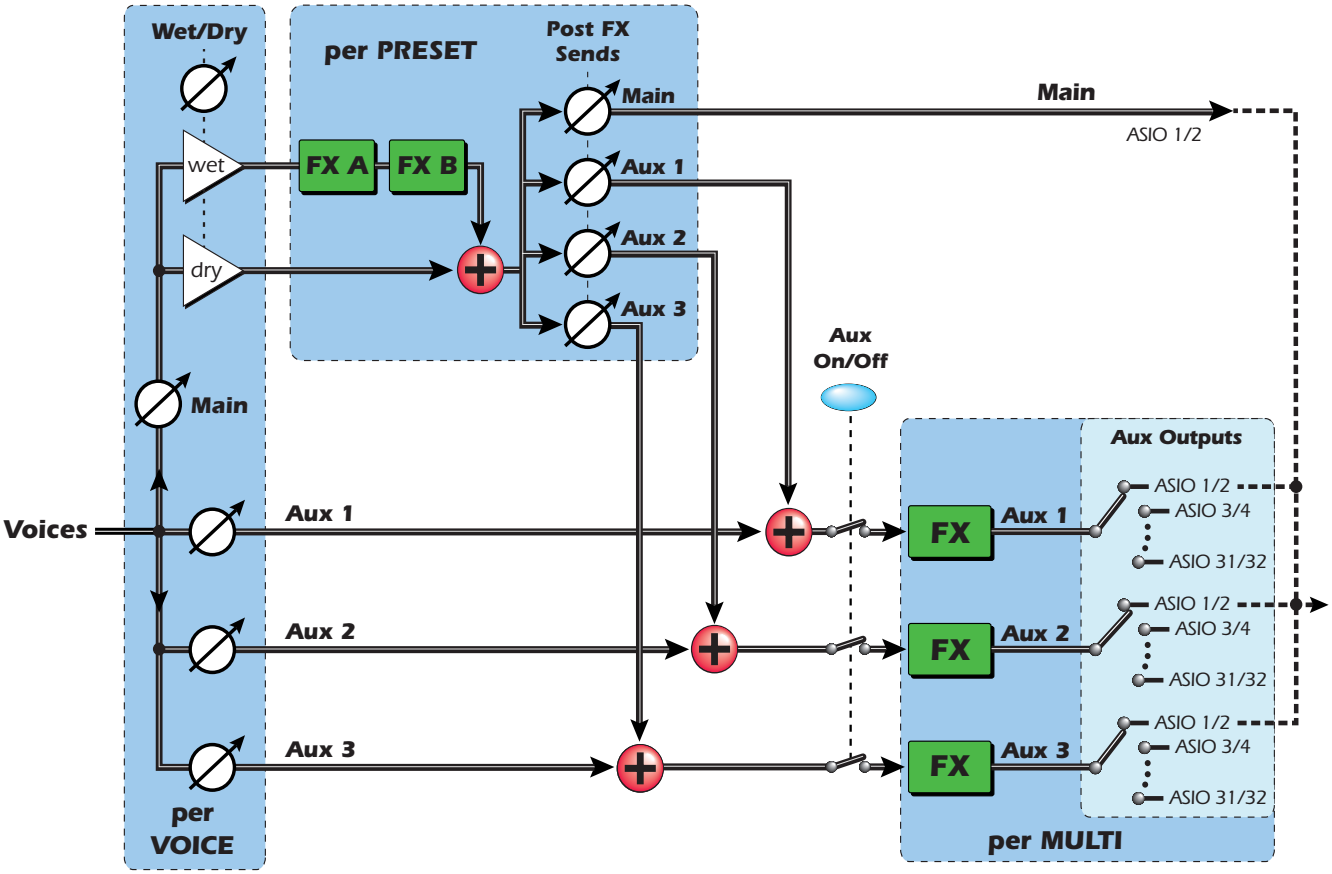
▼ Remember that if a preset with effects is used on more than one MIDI channel, the CPU penalty multiplies each time the preset is used.

▶ How to Temporarily Disable a Preset Effect

1. Select the **Multisetup Page** by clicking the Proteus 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.
4. **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

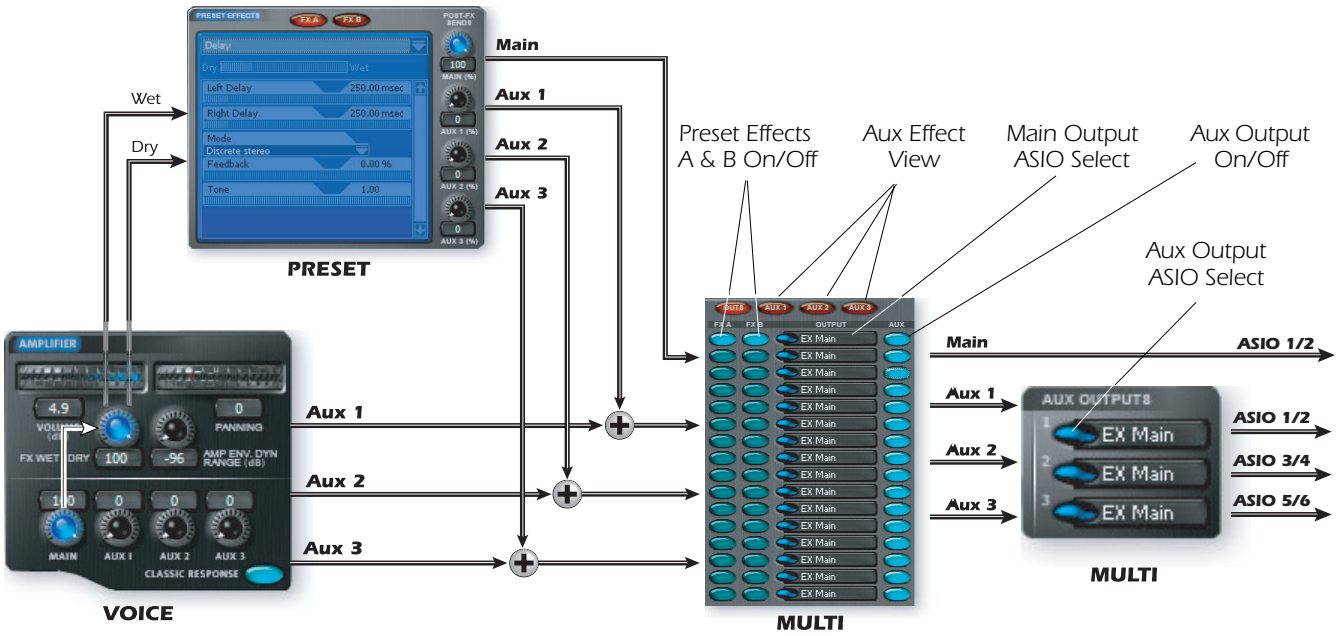


Two effects can be assigned per Preset. Three effects can be assigned per Multisetup. Effects amounts can be controlled at the voice level or the preset level. Note that since the Main output and Aux 1, 2 & 3 are all connected to the same ASIO output, they will be MIXED before being sent to the Host application (PatchMix DSP).

FX Routing

Make sure that you are thoroughly familiar with the architecture of Proteus X (page 23), 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 the Proteus 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 Proteus 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.

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.

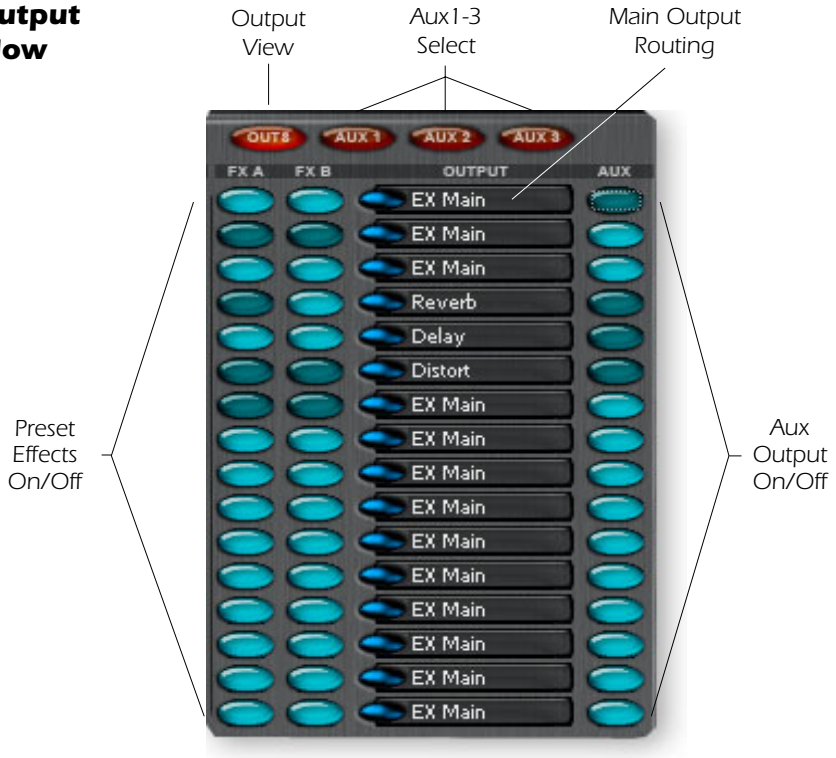
Main Output Window

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. *(Functions as an FX Bypass when the standard Proteus X Composer template is used. See page 108.)*

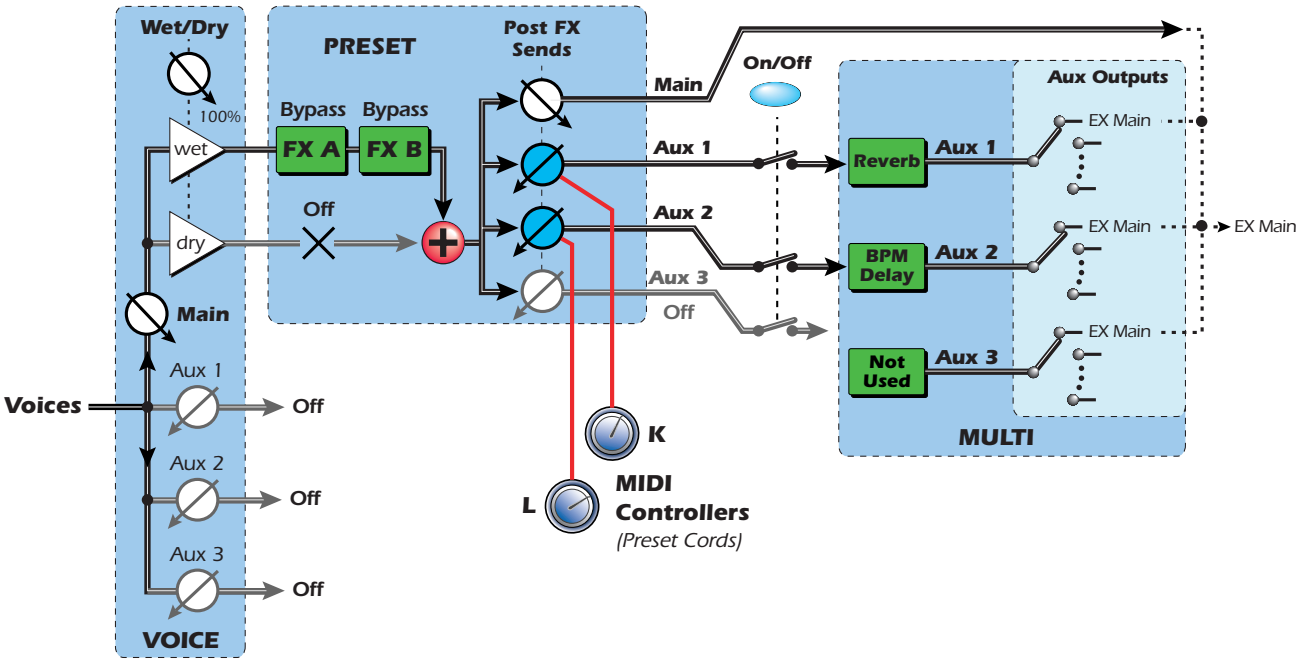
❖ ASIO channels must be created in PatchMix or another Host application before they appear in this list.

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 Patch-Cords 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 EX Main (ASIO 1/2) and so are mixed together before being sent to the Host computer.

Controlling the Preset Effect Amount from the Voice

The Proteus 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 many wonderful 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 that only occurs 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.

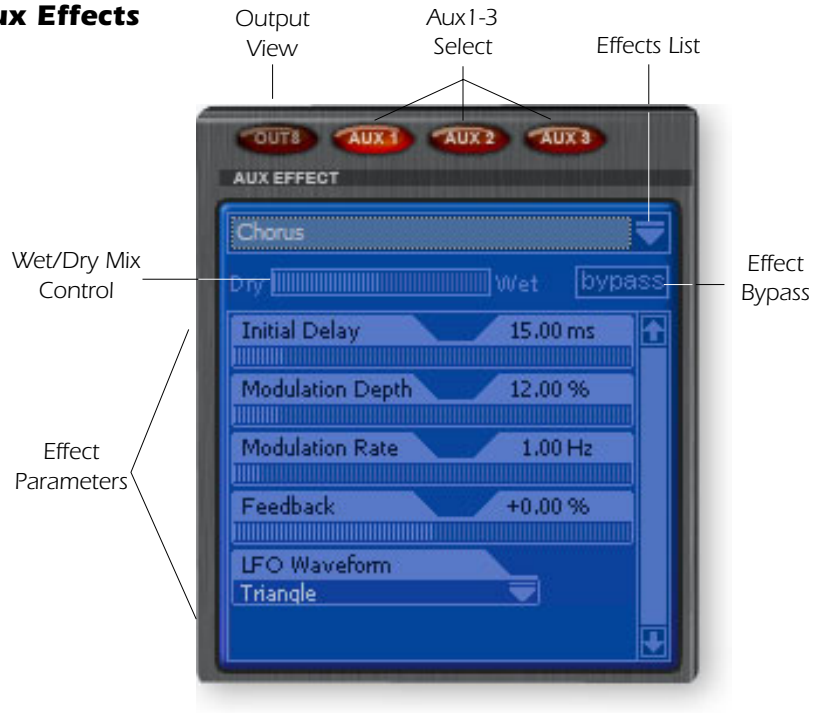
1. **Select the preset** you want in the tree by clicking on the keyboard icon. The Preset Global page appears. (Choose a sound with a fast release for this example.)
2. Click on the **Effect Select icon** at the top right corner of the Effects TV screen. A list of available effects drops down.
3. Select **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. Turn up the **Wet/Dry Mix** to about 86% Wet.
6. Select **Voice Processing** from the Tree for the preset you are using. The Voice Page appears.
7. Find an unused **Cord** and Set it up like this: **Velocity + | -100% | Wet/Dry Mix**
8. Make sure the **Wet/Dry Mix** in the Amplifier section is set to **100** (all wet).
9. **Play the keyboard** soft and hard to check it out.
10. 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.)
11. 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.

Aux Effects



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 uneffected 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. Refer to the FX Routing Diagram on page 105. Aux Effect parameters, such as Delay Time, Feedback, etc., can not be externally controlled.

Preset FX Screen

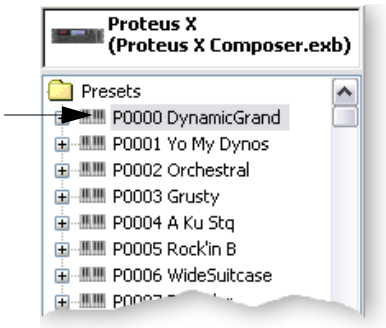
All effects have a wet/dry mix parameter to control the ratio of effect-to-plain signal, which is stored with the FX preset. Each effect has its own set of control parameters. Turn up the Wet/Dry mix to hear more effect while you adjust the parameters, then turn the amount down to the desired effect 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.

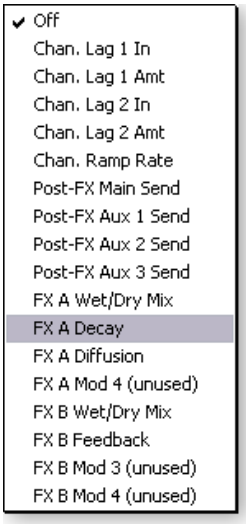
Preset FX Modulation Parameters

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 the Proteus X.

See page 21 to learn how to assign MIDI controller numbers to the Proteus's A-P reference letters.

Creating, Deleting & Reordering Templates

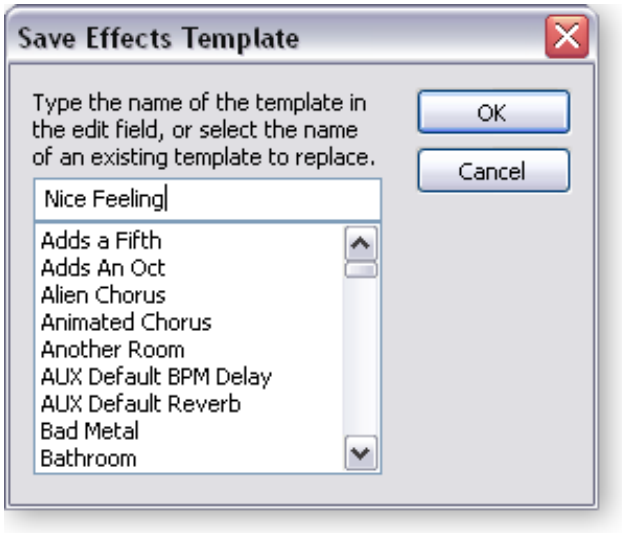
You can save a library of your favorite effects as *Templates*. Templates are saved with the Proteus X application so they're always available for use.

▶ To Create an Effect Template

1. Set the controls as desired for a particular effect.
2. **Right-click** with the mouse anywhere inside the TV screen border. A pop-up menu appears.
3. Select **Save Effects Template** or press **"S"** while the popup dialog box is visible. The following popup dialog box appears.

OR…

1. Select **Effects Templates** or press **"E"** from the Multisetup, Preset or Voice menus. A pop-up list of Templates appears.
2. Select **Save Effects Template** or press **"S"** while the popup dialog box is visible. The following popup dialog box appears.



3. Type in a name for your Template and click OK. The Template is saved with the editor program, ready to use when you open any bank.

▶ To Recall a Template

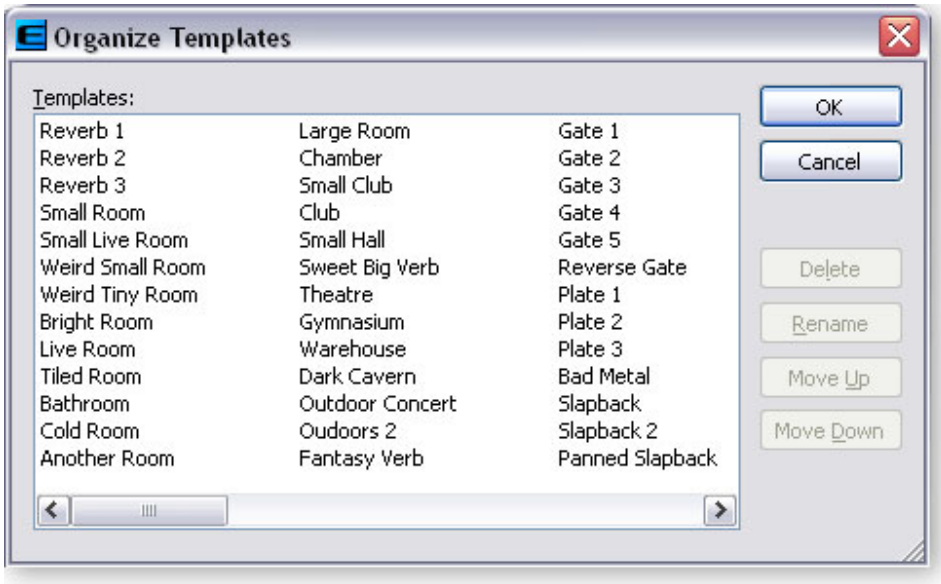
1. **Right-click** with the mouse anywhere inside the module border. A pop-up menu of Templates appears.
2. Select the desired Template from the list. The Template will be selected.

OR…

1. Select **Effects Templates** or press **"E"** from the Multisetup, Preset or Voice menus. A pop-up list of Templates appears.
2. **Select the Template** you want and click **OK**.

▶ To Rename or Delete a Template

1. **Right-click** with the mouse anywhere inside the module border.
2. Select **Organize Templates**. The following menu appears.



3. Click **Delete** to permanently remove the Template.
4. Click **Rename**, then type in the new name to rename the Template.
5. Click **OK** to close the dialog box.

▶ To Change the Order of the Templates

1. **Right-click** with the mouse anywhere inside the module border.
2. Select **Organize Templates**. The menu shown above appears.
3. Click **Move Up** to move the Template one space up in the list.
4. Click **Move Down** to move the Template one space down in the list.
5. Click **OK** to close the dialog box.

Effects Listing

Reverb	Delay *(mono)*	Ring Modulator
Early Reflections Reverb	Flanger	SP12-ulator
Reverb Lite *(mono)*	Phaser	Growl
Chorus	Early Reflections	Tube
Chorus / Delay *(mono)*	EQ - 1Band Parametric	Twin *(mono)*
Delay	EQ - 4 Band	
Delay (BPM)	Pitch Shifter *(mono)*	

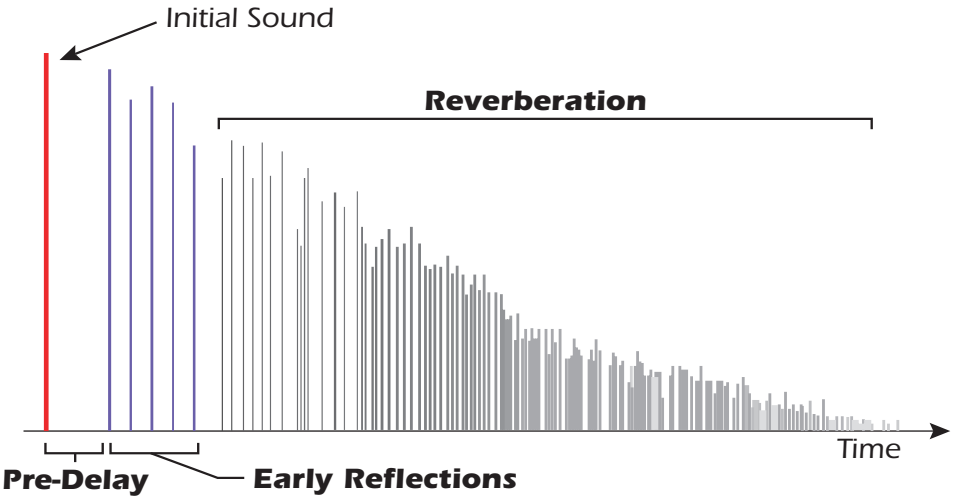
Effects Descriptions

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:1millisecond 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	**Independent -** In this mode, the left and right sides function as completely separate reverb units. **Figure Eight -** This mode incorporates criss-cross feedback in which the left channel feeds the right and vice-versa. **Matrix -** This mode utilizes multiple feedback paths to create a spacious, natural sound.

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 115.

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, Multitap, 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	**Independent -** In this mode, the left and right sides function as completely separate reverb units. **Figure Eight -** This mode incorporates criss-cross feedback in which the left channel feeds the right and vice-versa. **Matrix -** This mode utilizes multiple feedback paths to create a spacious, natural sound.

■ 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 horse-power. See also "Background: Reverb" on page 115.

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	Sets how much the LFO affects 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

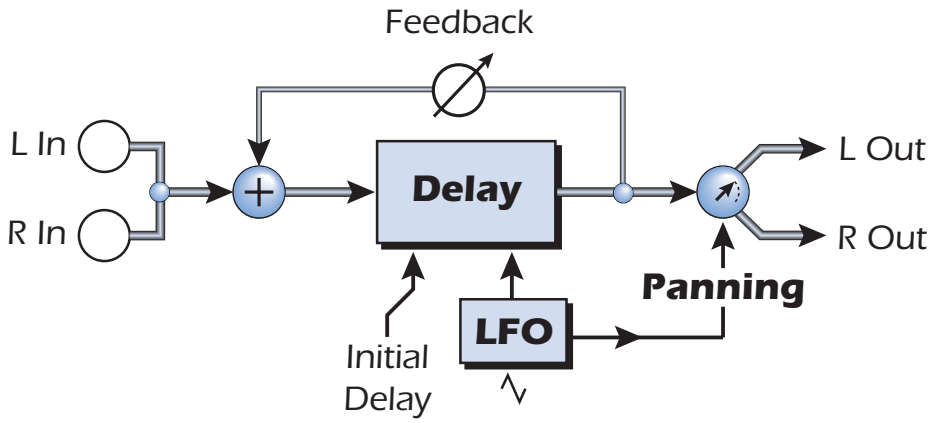
Chorus / Delay

This effect is a combination Chorus and Delay because the signal can be delayed up to a full second (1000ms). Slightly modulating the delay time in the 15-20ms range produces the familiar chorus effect. See "Background: Chorus" on page 118.

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.

❖ *Very short delay times combined with a high feedback amount can be used to create monotone robotic-sounding effects.*

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.



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

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 126.

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.

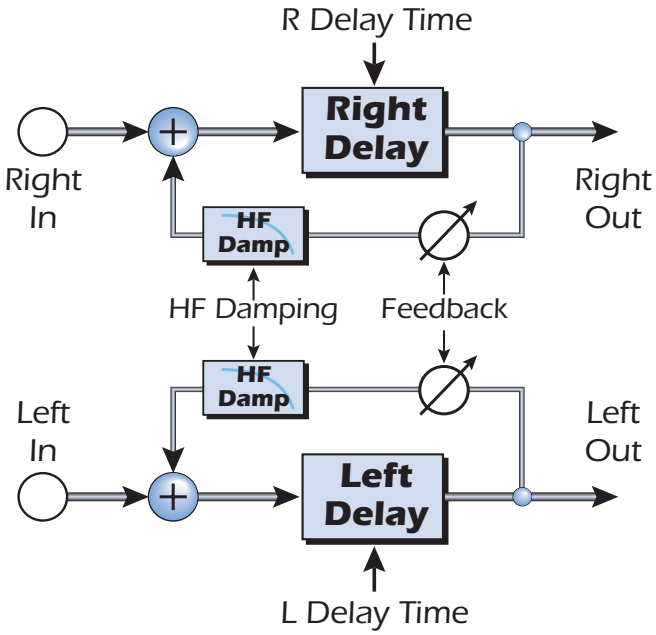
Discrete stereo

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.



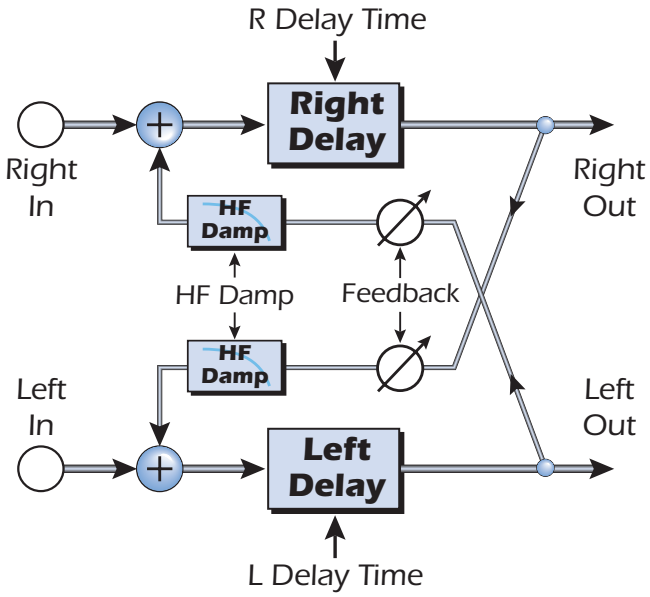
Ping Pong, stereo

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.



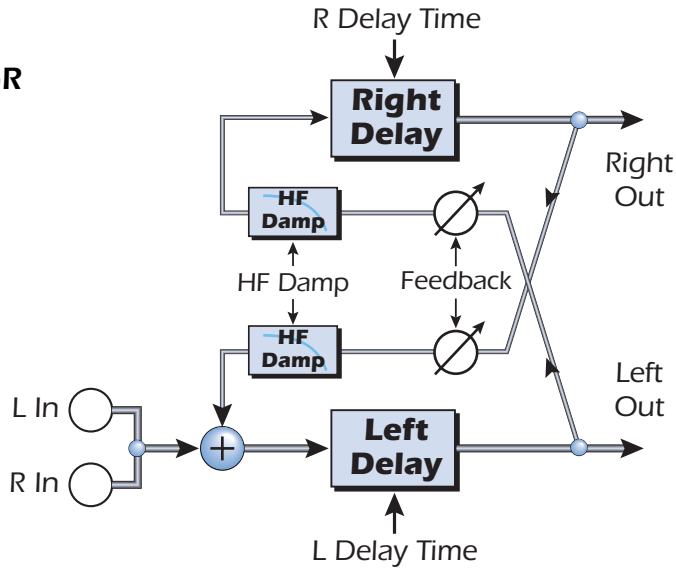
Ping Pong, mono L->R

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

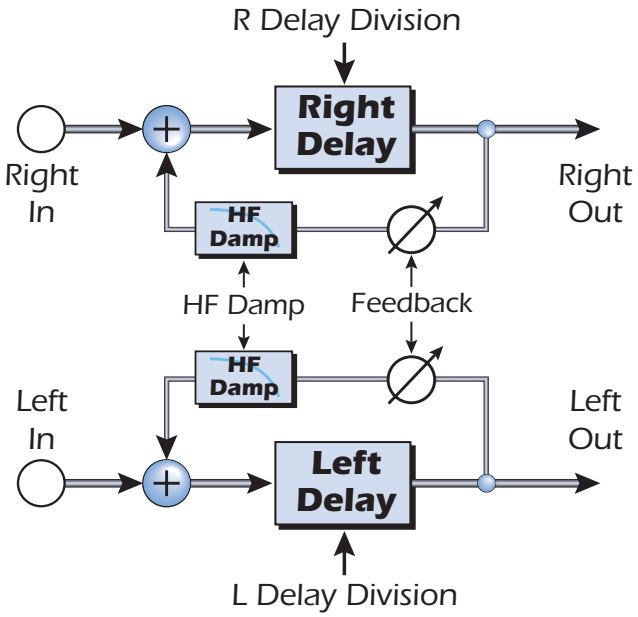
Delay (BPM)

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 120 for full descriptions and diagrams. The BPM Tempo Chart on page 83 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 in milliseconds. Range: 4/1 note to 1/64 dotted-note
Right Delay Division	Sets the length of the delay for the right channel in milliseconds. 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

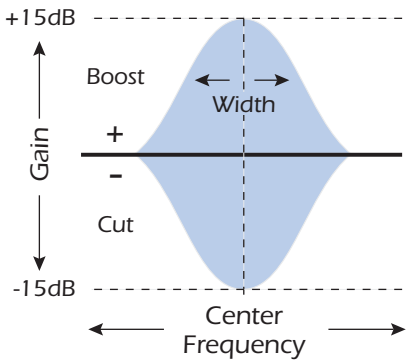
Early reflections are the initial echoes that you hear when listening to sound in an indoor space. The echoes are staggered because they are reflected off surfaces which are at various distances from your ear. Normally this effect would be part of a reverb algorithm, but we decided to include it as a 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 just want to brighten up the lead vocal a bit, you might choose this EQ. 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

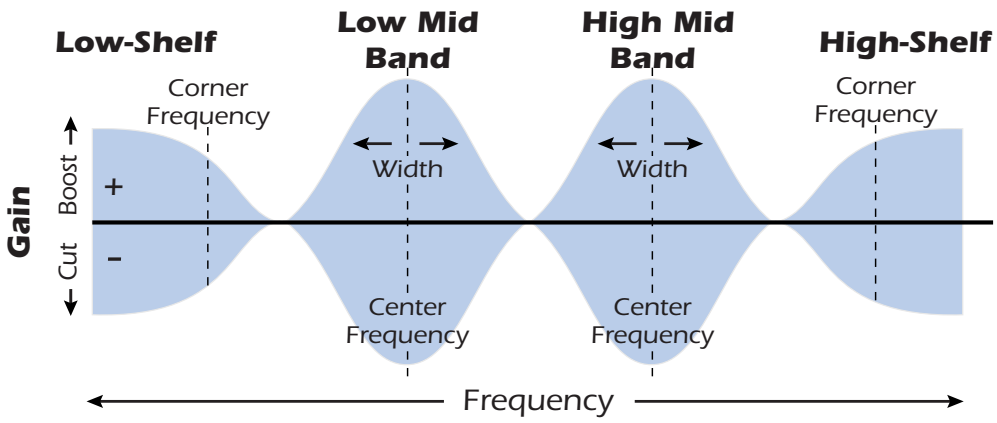
Modulation Parameters

- 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.



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

Note: The Wet/Dry Mix control on an equalizer should normally be set to 100% wet or unpredictable results may occur.

Tip: When using a MIDI controller to change the EQ gain, a value of 77 = 0dB.

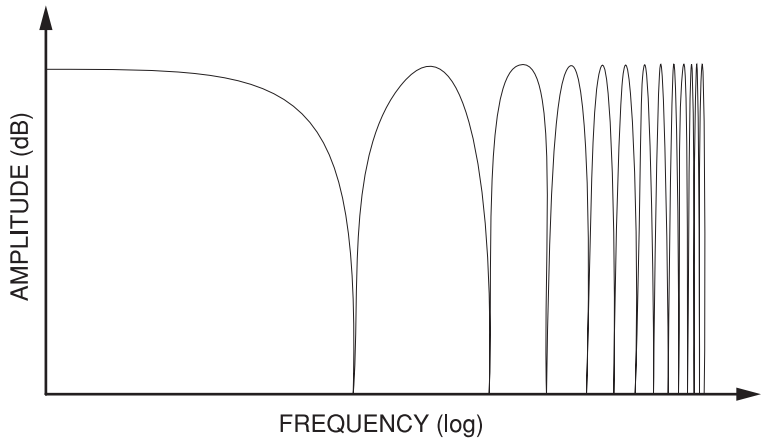
Modulation Parameters

- 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.

❖ 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).



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.

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 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 0% to 100%
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

Growl

This is a new kind of self-modulation effect created especially for the Proteus 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 Grrrreat!

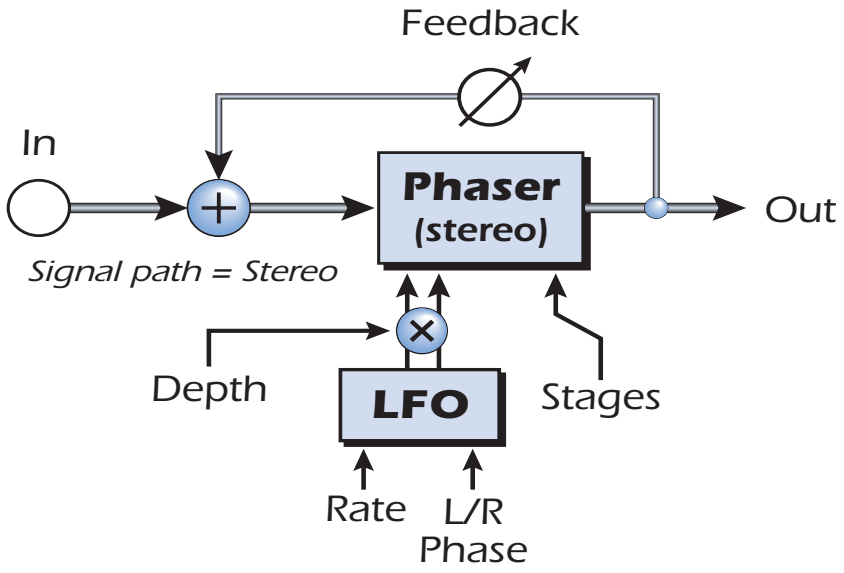
Parameter	Description
Initial	This control "Tunes" the effect to a specific frequency range. Range: .01ms to 15ms
Depth	Sets how much self modulation is applied. Range 0% to 100%
Color	Controls the tonal color of the effect. Range: -99% to +99%
Pre-Filter	Anti-aliasing filter. Turn this filter up if you hear aliasing at high-frequencies.

Modulation Parameters

- Wet/Dry Mix
- Depth
- Color

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 animation to the sound.

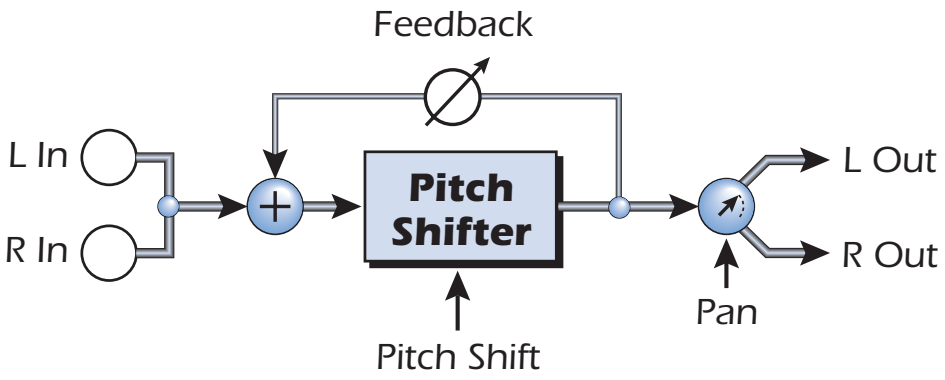
Modulation Parameters

- 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 Proteus X keyboard.) The range of the pitch shifter is -36 semitones down to +24 semitones 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 semitones to +24 semitones
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

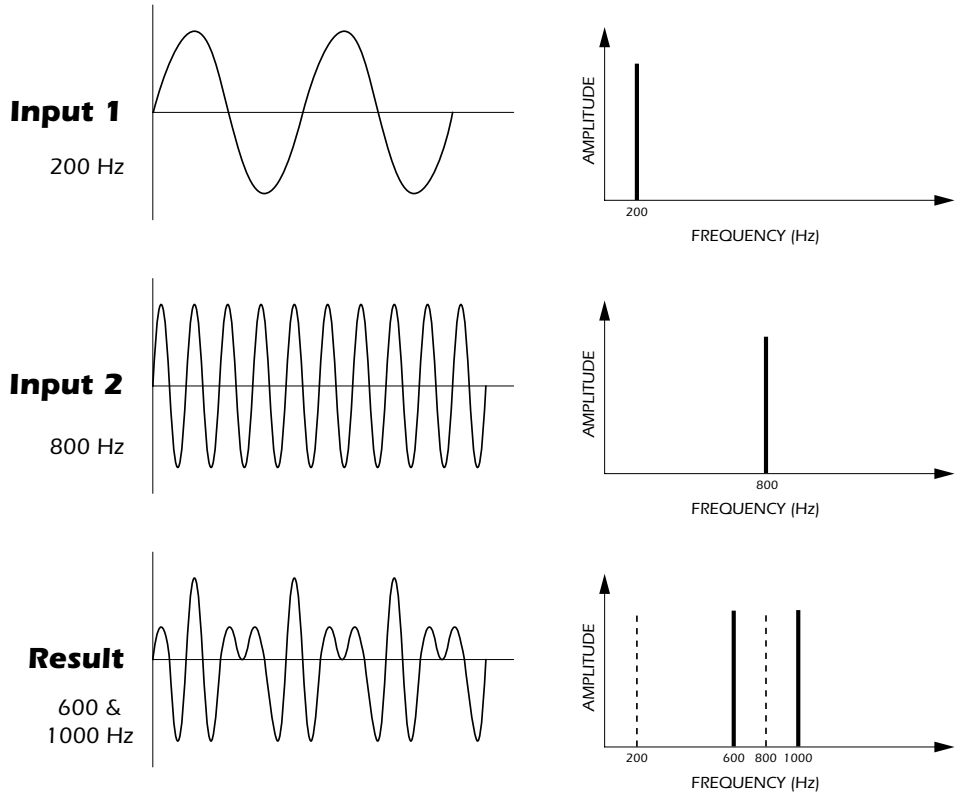
- Wet/Dry Mix
- Pitch
- Feedback
- Pan

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 harmonic of each signal multiplies every other according to its amplitude. Because ring modulation is a linear process, these new sum and difference frequencies are displaced 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.

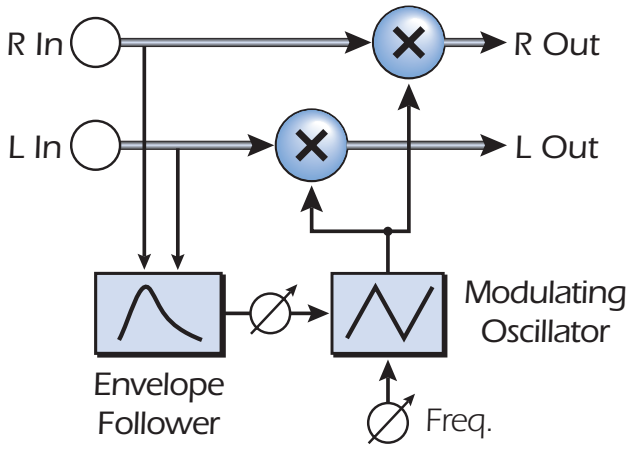
The envelope follower sweeps the pitch of the modulating oscillator up or down based on the amplitude of the input signal. Try adjusting the Proteus X attack and decay controls (amp) to hear the effect of the envelope follower. The envelope follower can sweep the modulating oscillator in either the positive 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

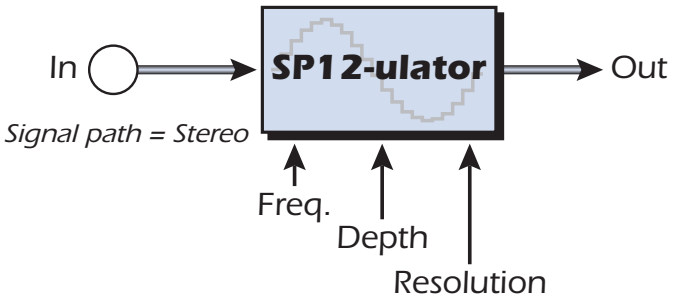


Modulation Parameters

- 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 remains a classic. The SP-12 used a gritty, "drop-sample" pitch shifting technique which contributed to its distinctive sound. The SP12-ulator effect pays homage to the venerable SP-12, and allows you to mangle sounds as your forefathers did many years ago.



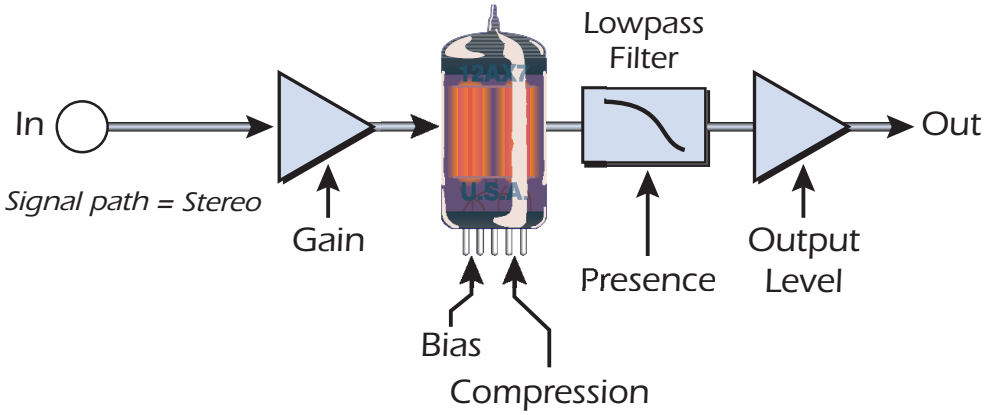
Parameter	Description
Frequency	This parameter controls the number of samples that will be "dropped." Range: 0 - 100
Depth	This parameter controls the intensity of the resultant distortion. Range: 0% to 100%
Resolution	The SP-12 is a 12-bit drum machine. The SP12-ulator allows you to reduce the bit depth 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

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.



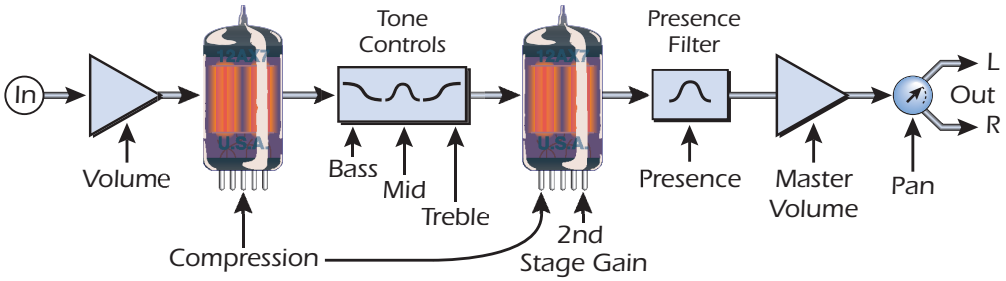
Parameter	Description
Gain	This parameter controls the overdrive amount on the “tube” and thus, the amount of distortion generated. Range: 0.25 - 50.00
Output Level	This parameter sets the output volume level and has no effect on the tone.
Presence	This control sets the lowpass filter frequency at the 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 timbrel 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

8 - Controls

This chapter describes how to work the controls of the Proteus X. There are certain tips and tricks to learn in becoming a power user, and a good number of them are in this chapter.

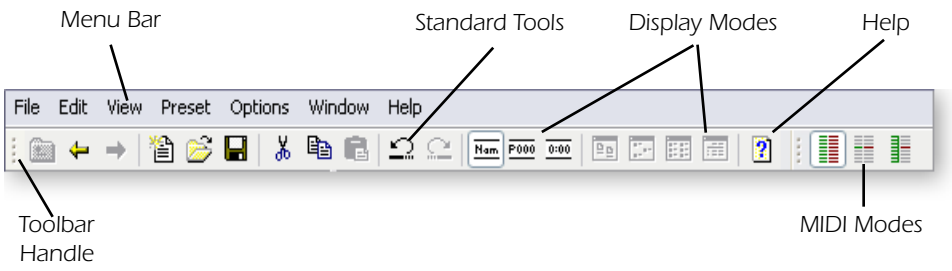
The Toolbars

The toolbars are located just beneath the menu bar at the top of the window and one more at the very bottom of the window (the status toolbar). 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.

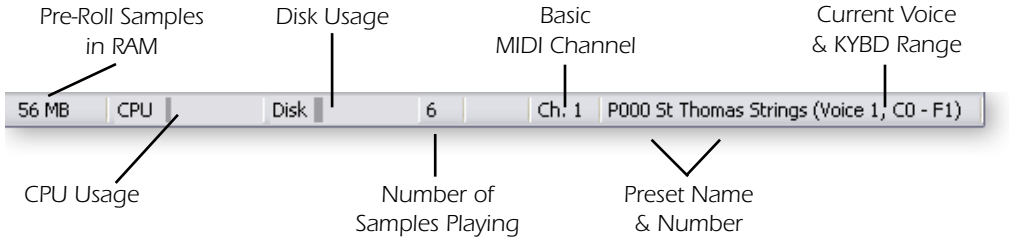
The upper toolbars can be rearranged by simply clicking the Toolbar Handles (see below) and dragging them wherever you want.

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

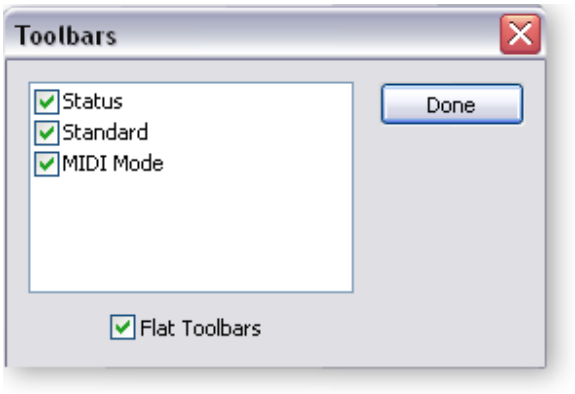


Status Toolbar



Viewing and Hiding the Toolbars

Select **Toolbars** from the **View** menu to call up the Show/Hide Toolbar menu. Check the boxes of the toolbars you wish to view. "Flat Toolbars" changes the look of the toolbars.



Drag & Drop

You can use simple "Drag & Drop" operations for many Proteus 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. Page 173 provides a complete listing of the drag & drop functions.

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 cursor is over a field.
- **Typing** the desired value into the field.

▶ Move the Cursor from the 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:

- Use the **Left/Right** keys to increment or decrement the value by 1.
- Use the **Page Up** & **Page Down** keys increment or decrement by 10.
- 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 Key Ranges & Fades

The Key map windows contain alphanumeric fields displaying the key ranges of samples and voices.

▶ Editing the Key Range using the Graphic Display:

- *Click and drag the end points of the graphic display to set the high and low key range.*
- *Hold Ctrl, then click and drag the end points to move the fade range.*

▶ Editing the Key Range via the Alphanumeric Fields:

First, 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**. *A small MIDI connector symbol appears at the bottom of the window.* *Play your MIDI keyboard to set the key position.*

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 a MIDI icon 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.

Voice Processing Graphic Keyboard Operation

The Voice Processing window has a small keyboard at the top. Clicking the keyboard will play the clicked note. The range bar below the keyboard shows the range of all selected voices. You can change the voice selection by simply dragging the ends of the bar.

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 81.

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 54). This allows the synthesizer to adapt to the tempo of your music.

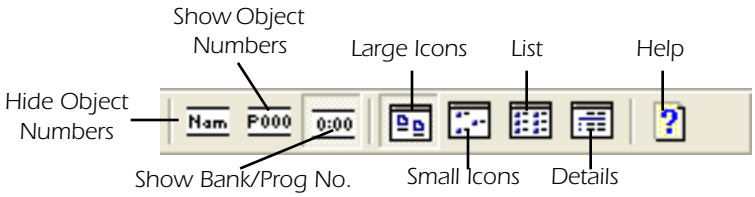
The tempo automatically syncs to an external MIDI clock if it is present in the system. If you have an external MIDI device generating MIDI clock, or if another application is generating MIDI clock, the tempo will automatically lock to the MIDI clock tempo. If nothing else in the system is generating MIDI clock, the internal clock will be used.

Cut, Copy & Paste

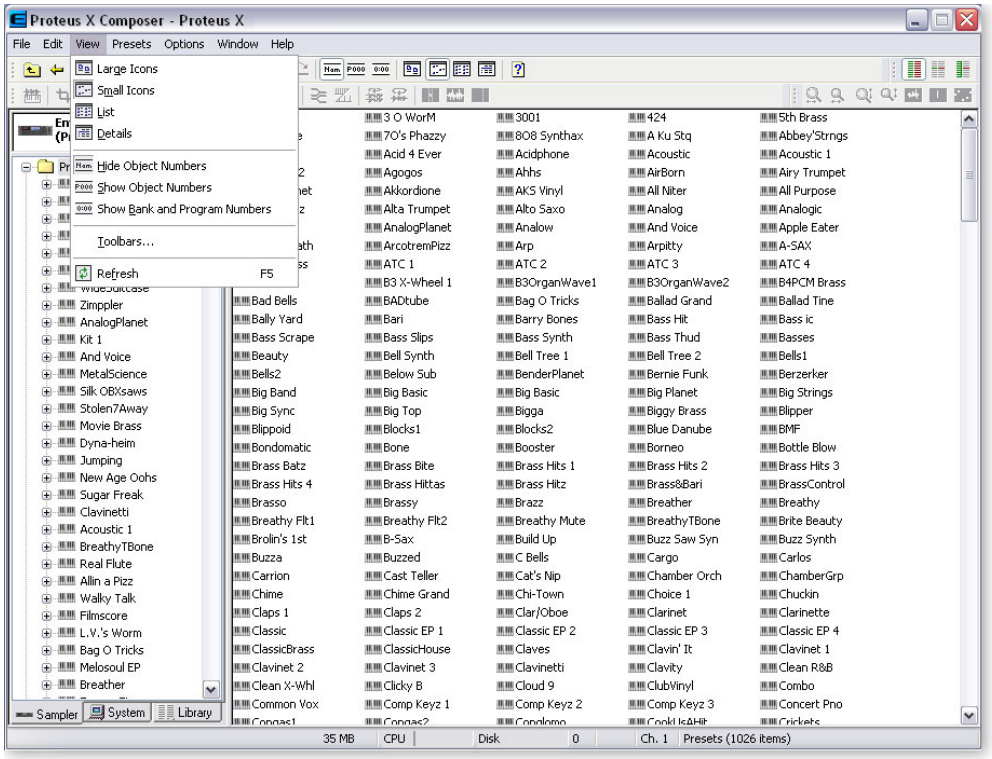
You can use cut, copy and paste commands for most Proteus X functions.

Page	Uses
Voices & Zones	▪ Cut, Copy and Paste voices between banks. ▪ Rearrange the order of voices in the list.
Voice Processing	▪ 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



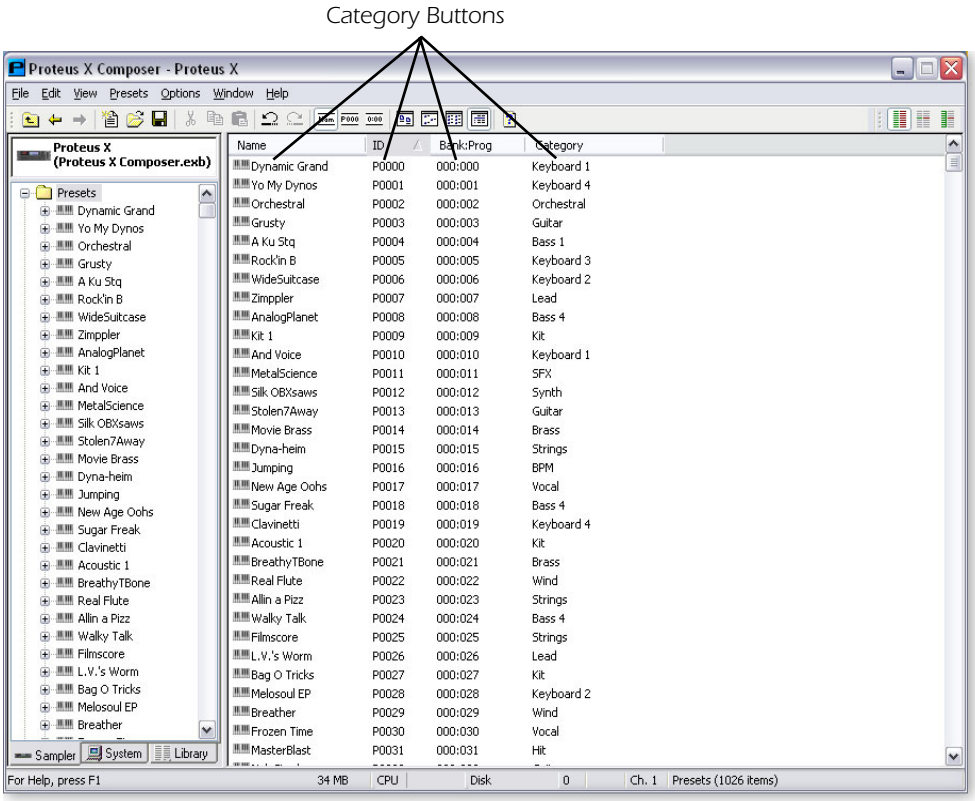
Toolbar View Selection



Main Display View

When in the Sample, Preset or Multisetup Tree, clicking on the sample or preset folder shows all the samples or presets in the bank in the large window. The **View Menu** provides several display options for samples, presets and multisetups. 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	*Displays the ID Number, Bank-Program Number, and Category. Clicking on the Category button at the top of each column sorts the list by that category.*



Tree View

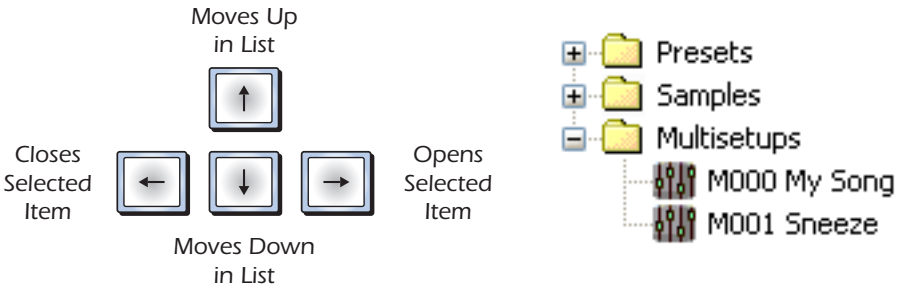
The Sample, Preset or Multisetup Tree items can also be viewed in several ways. The **View Menu** provides several display options for the tree. The view options are listed below:

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).

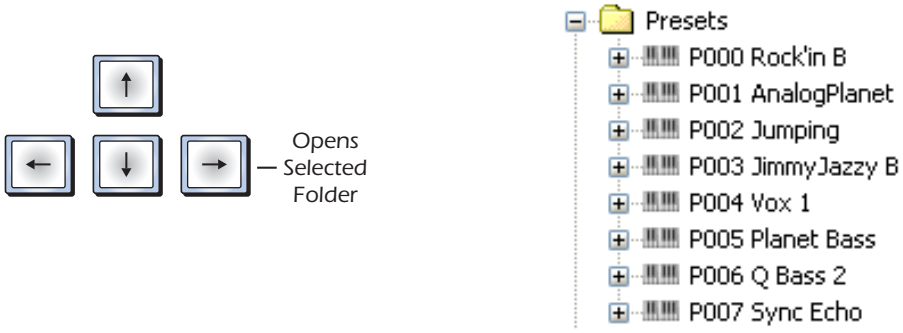
Tree Navigation using the Keyboard

Cursor Keys

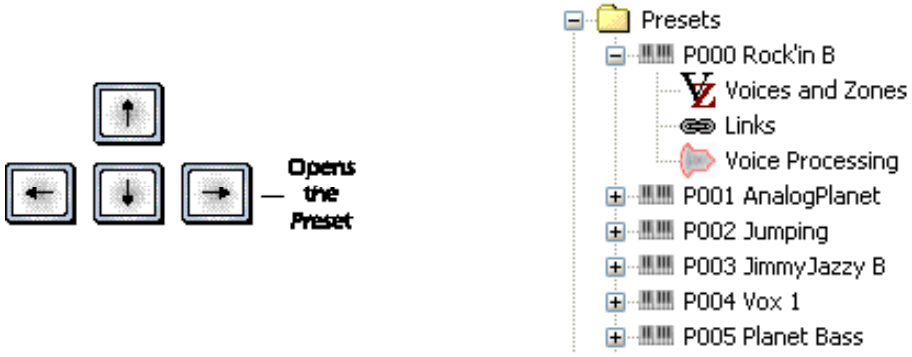
You can move quickly through the Tree using the Left-Right-Up-Down Cursor Keys.



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.

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 the Proteus X. Simply right-click over the field or area to bring up the menu for that function. Please refer to page 172 for a complete listing of context-sensitive menus.

Undo/Redo

Almost every action you do with the Proteus X can be undone. You can undo up to 100 actions. Select Undo or Redo from the Edit menu or press Ctrl+Z (Undo) or Ctrl+Y (Redo) from the keyboard.

9 - 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 a also a rich source of information on this subject. Seek and you shall find.

Your initial involvement with the Proteus 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.

Editing Presets

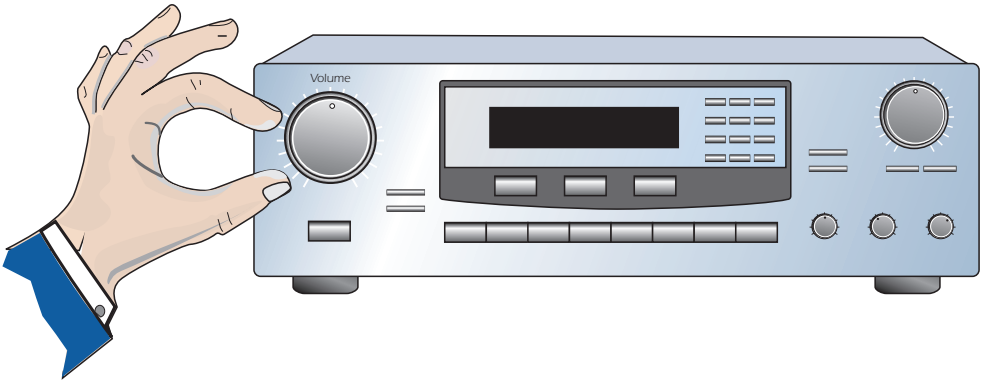
It's easy to create new presets by modifying existing presets. This is really the best way of getting acquainted with the Proteus X. If you don't like the results, simply reload the preset or bank and you'll be back to the original sound. Changes are not made permanent until you SAVE a bank. Therefore, 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.

The Proteus 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.

The Proteus 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 21).

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.

Noise & Random Generators

Generate spectrums of noise and random signals to be used for modulation.

T-switch and Footswitches

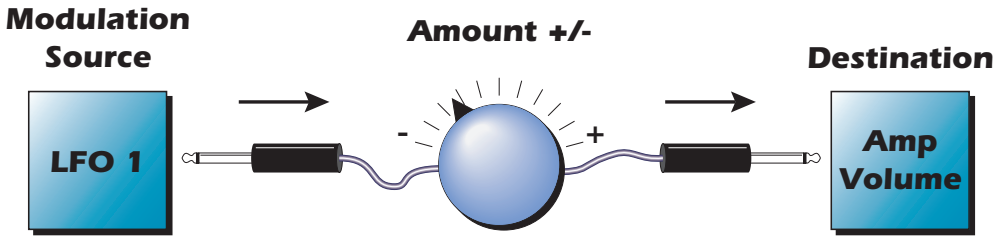
Change a parameter when a MIDI switch is pressed. MIDI footswitch numbers are programmed in the Preferences dialog box (See page 21).

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.



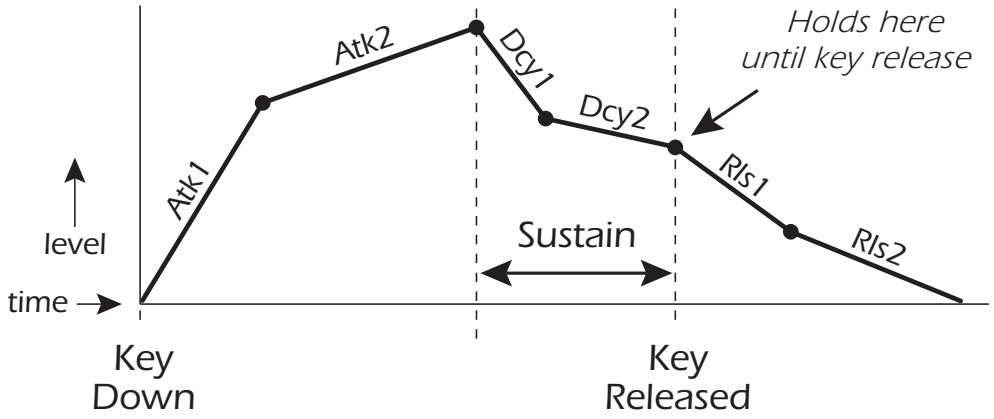
Envelope Generators

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 EOS envelope generators are just like standard ADSR's, 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.



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.

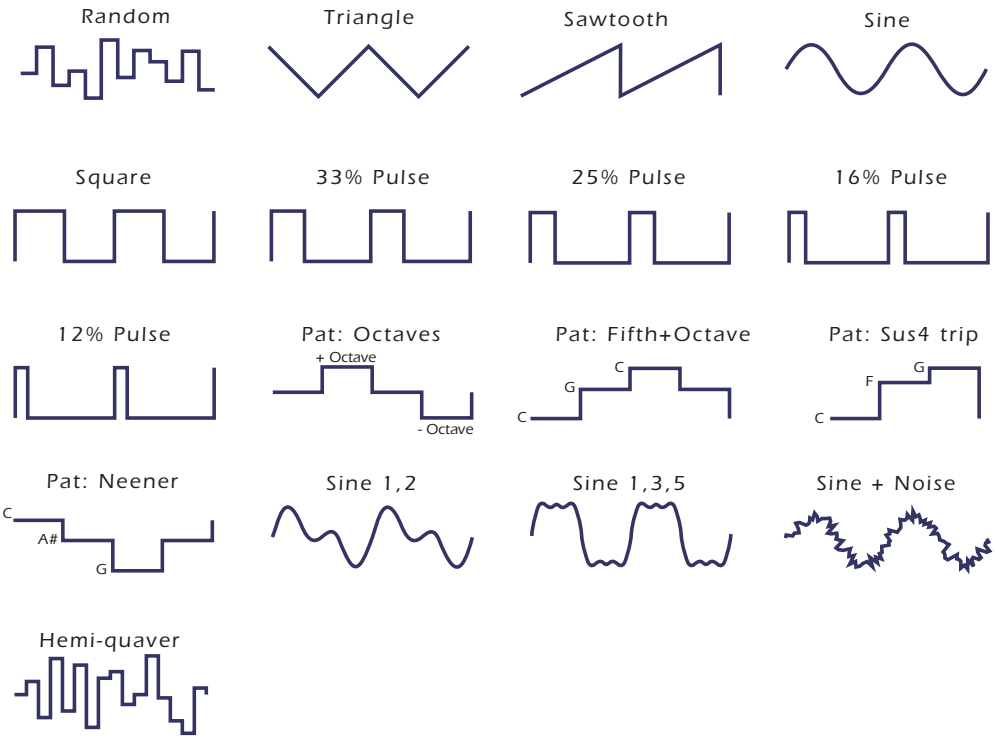
❖ By routing the Auxiliary Envelope to control the pitch (Cords) you can easily hear the shape of the envelopes you are creating.

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 148.) 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.

Low Frequency Oscillators (LFOs)

A Low Frequency Oscillator or LFO is simply a wave which repeats at a slow rate. The Proteus has two multi-wave LFOs for each channel. The LFO waveforms are shown in the following illustration.

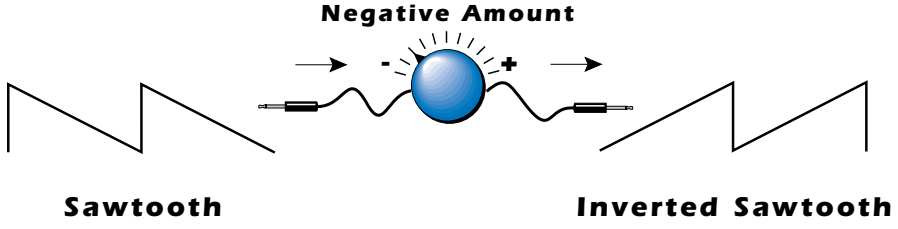


By examining the diagram of the LFO waveforms, you can see how the LFO will affect a modulation destination. Suppose we are modulating the pitch of an instrument. The sine wave looks smooth, and will smoothly change the pitch. The square wave changes abruptly, and will abruptly change the pitch from 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.

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

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 -> 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 LFO 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 152 for information on how to use a DC Cord.

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.

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 "synchro-sonic" 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.

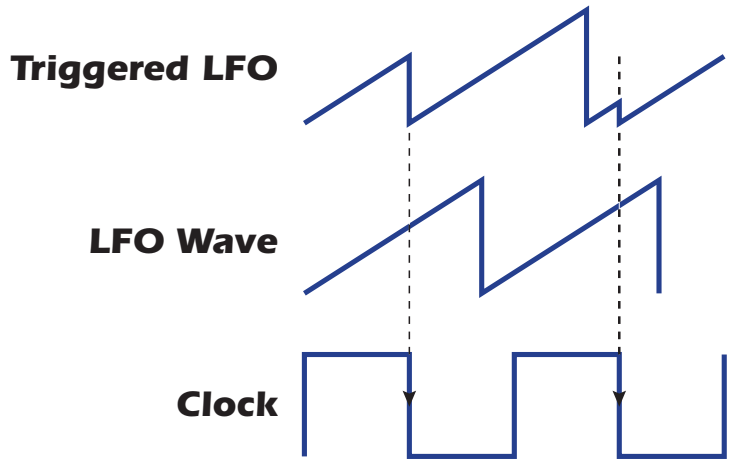
- *Envelopes are triggered on the positive going edge of the clock. LFOs are triggered on the negative going edge of the clock.*

- *The tempo of the master clock is set from the Multisetup page.*

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.

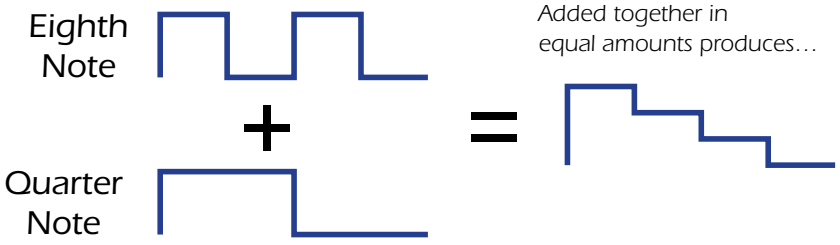
If the LFO rate is close to the clock rate, **the LFO will synchronize with the clock.** If the two rates are far apart, the waveform of the LFO will be mildly or radically altered (as shown by the diagram on the following page).

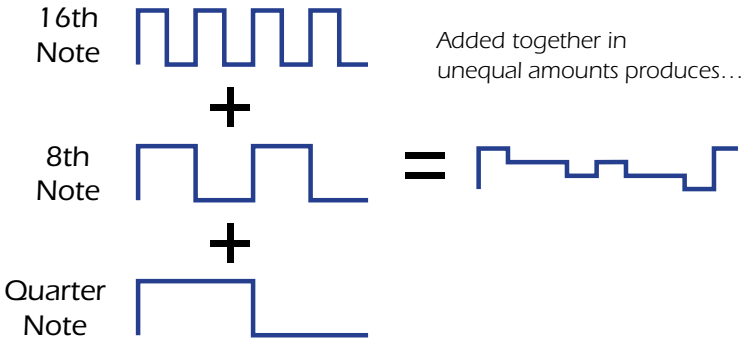


LFO Trigger causes the LFO to reset each time the clock waveform goes low.

The possibilities of clock modulation and retrigger are numerous and varied. 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.

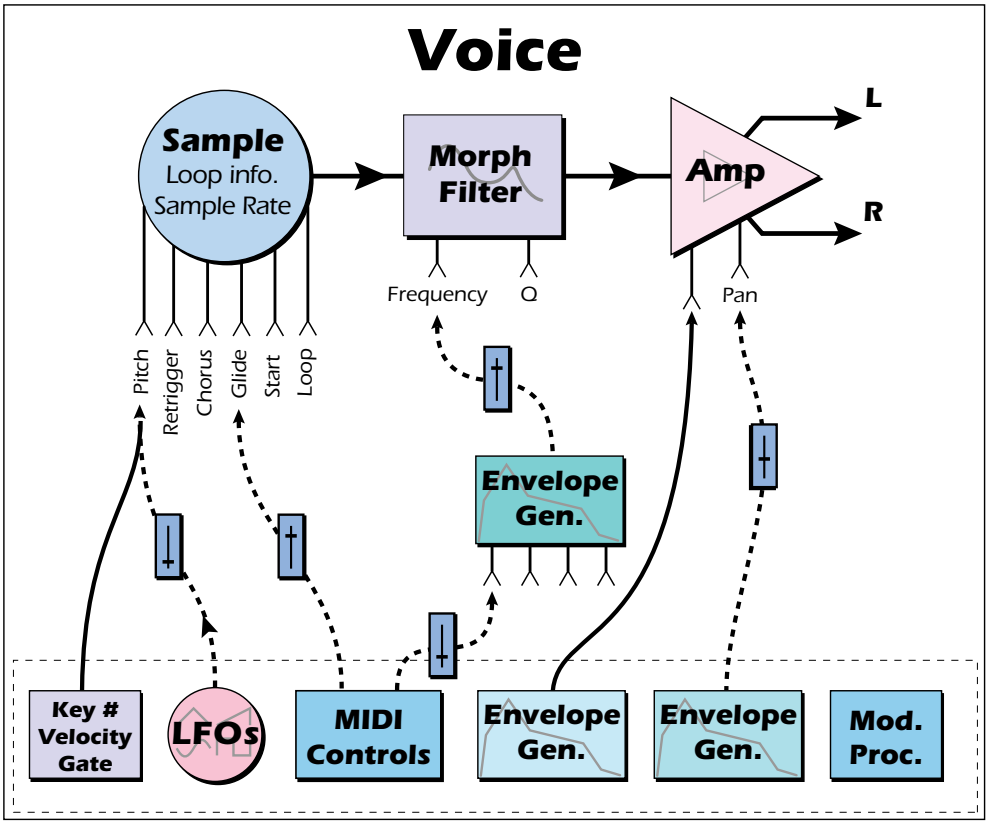
- *Turn different voice layers on and off using different clock divisors.*
- *Create a global triangle LFO by routing the Clock through a Lag Processor.*
- *Switch between Auxiliary and Filter Envelope retriggering using a slider or footswitch.*
- *Retrigger LFOs or Envelopes using noise or other LFOs 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.)*





Modulation Destinations

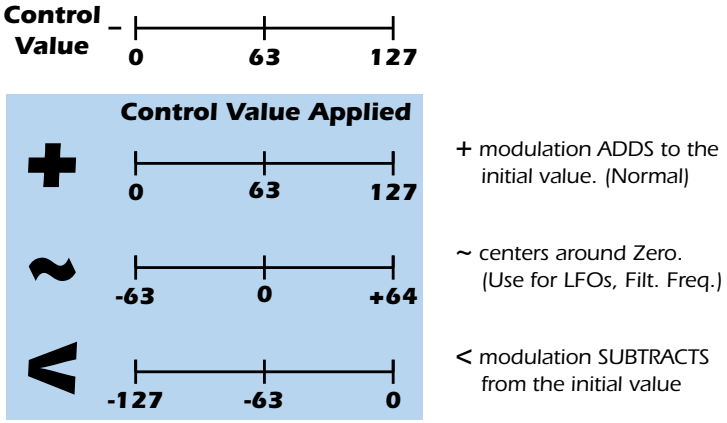
The Cords section of the Preset Edit module is where you connect sources to destinations. Each cord has an amount control associated with it to control how much modulation is applied.



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.



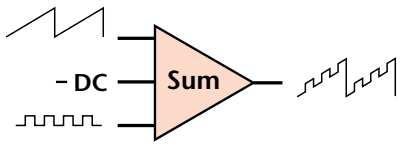
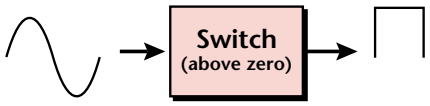
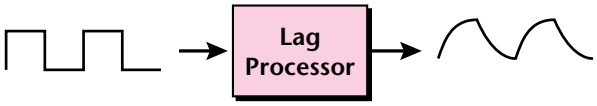
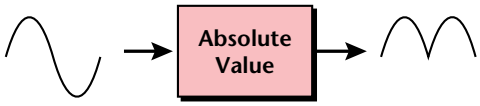
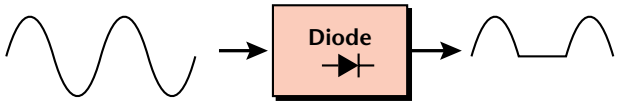
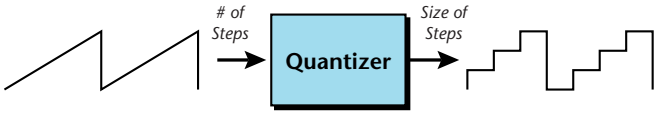
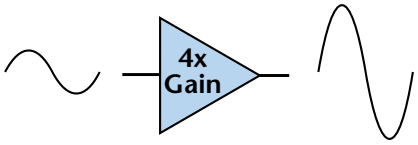
+ modulation ADDS to the initial value. (Normal)

~ centers around Zero. (Use for LFOs, Filt. Freq.)

< modulation SUBTRACTS from the initial value

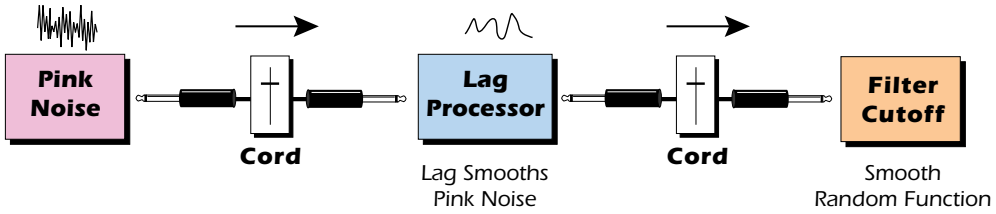
Modulation Processors

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:

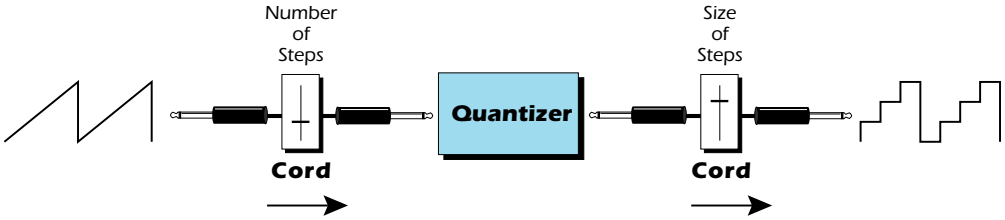
Summing Amp Allows you to add several modulation sources so the output can be controlled by a single Cord Amount.	
Switch Outputs a full value (100) level whenever the input is greater than zero.	
Lag Processors Slow down rapid changes in the input signal. The output "lags" behind the input. Rate is adjustable.	
Absolute Value Inverts negative values and leaves positive values untouched. Outputs only positive values.	
Diode Blocks negative values and allows positive values to pass untouched.	
Quantizer With the input Cord set to 100% the output is limited to 16 discrete steps.	
4x Gain This processor multiplies the input value by a factor of four. Used to amplify modulation sources.	
Flip-Flop Alternates between full value (100) and 0 each time the input goes positive through zero.	

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.



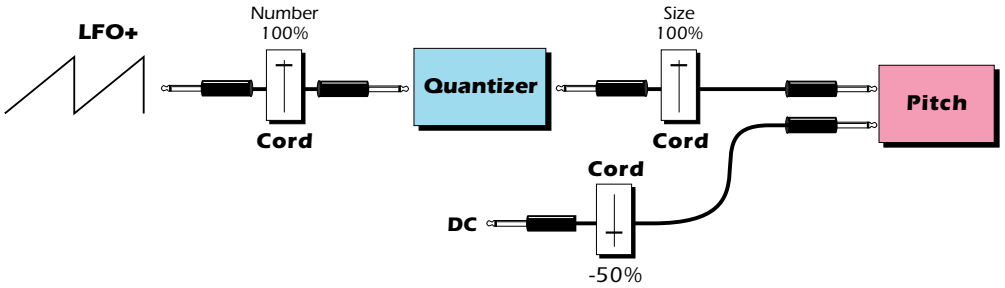
The Quantizer can generate interesting whole-tone scales when envelope generators or LFOs are routed to the input. The quantizer turns a smoothly changing input signal into a series of steps. By routing the output of the quantizer to Pitch and adjusting the cord amounts, you can control both the number of steps and the pitch interval of each step.



The input cord amount controls how many steps are generated. 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.)

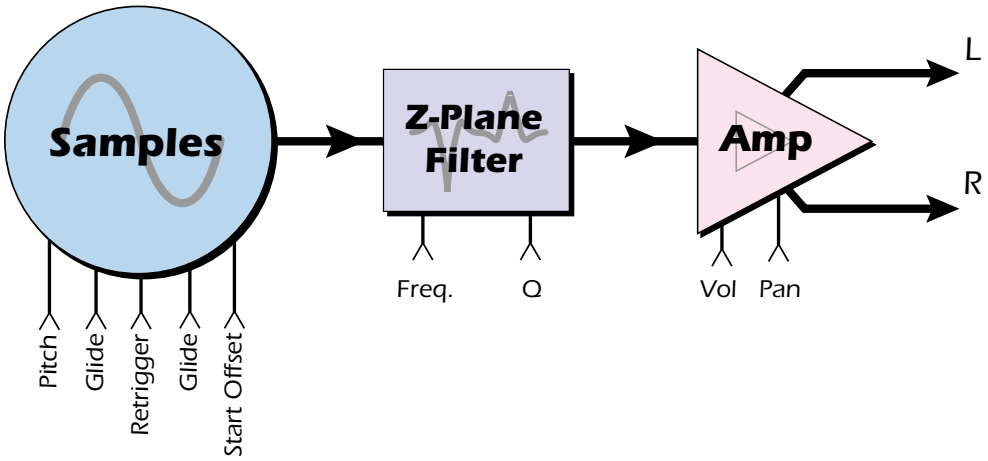
❖ Experiment with this patch by connecting other sources and destinations to the quantizer.



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

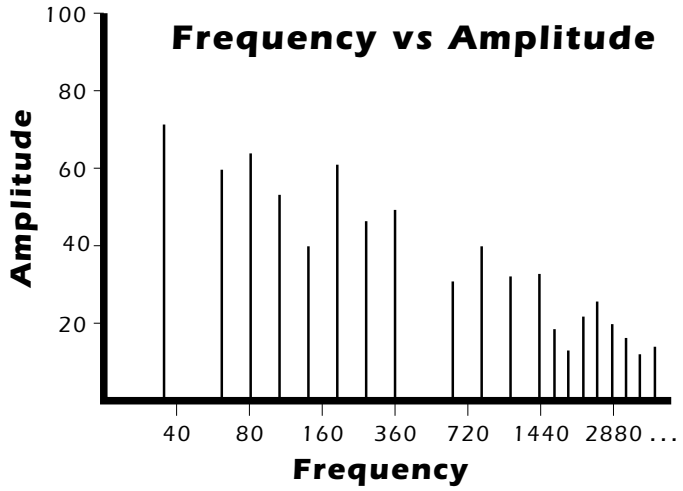
Following is a block diagram of the signal path.



The Proteus X utilizes advanced Z-plane filters which can dramatically alter the sound of a sample.

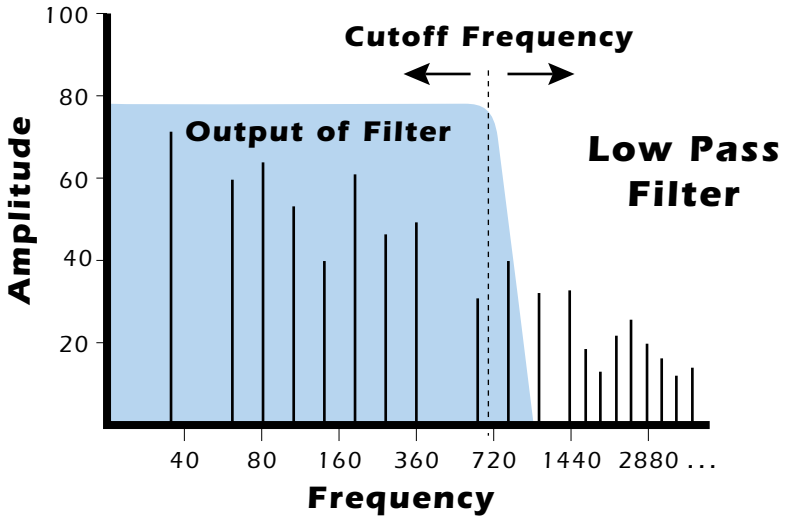
To understand how a filter works, we need to understand what makes a sound wave. A sine wave is the simplest form of sound wave. Any waveform except a sine wave can be analyzed as a mixture of sine waves.

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.

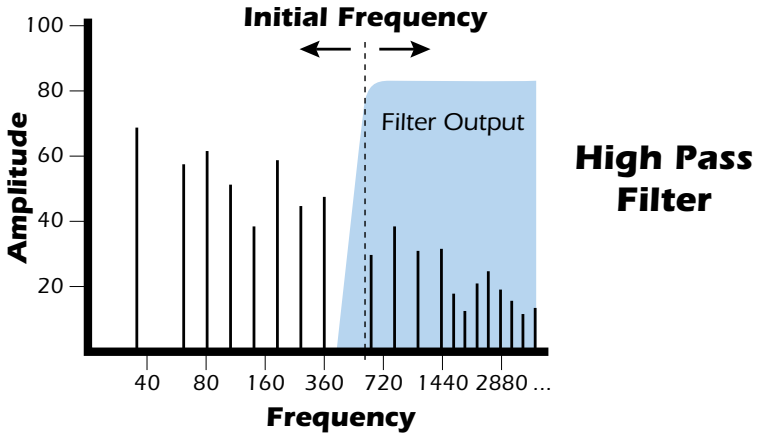


What is a Filter?

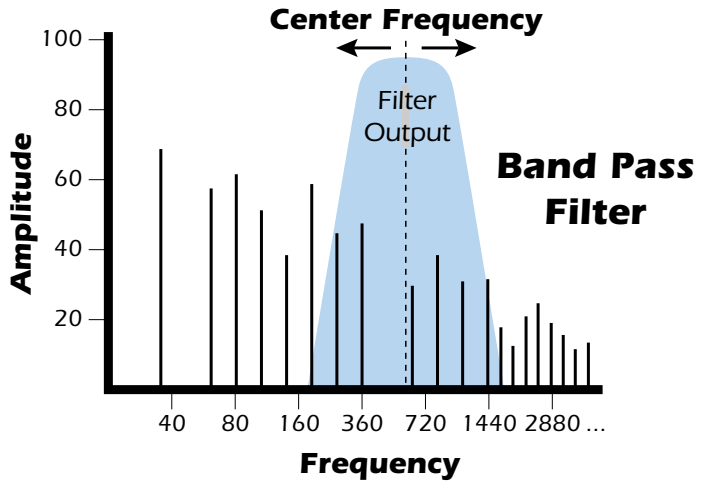
Most samples are complex waves containing many sine waves of various amplitudes and frequencies. A filter is a device which allows us to remove certain components of a sound depending on its frequency. For example, a Low Pass Filter lets the *low frequencies pass* and removes only the high frequencies.



A filter that lets only the high frequencies pass is called a high-pass filter.

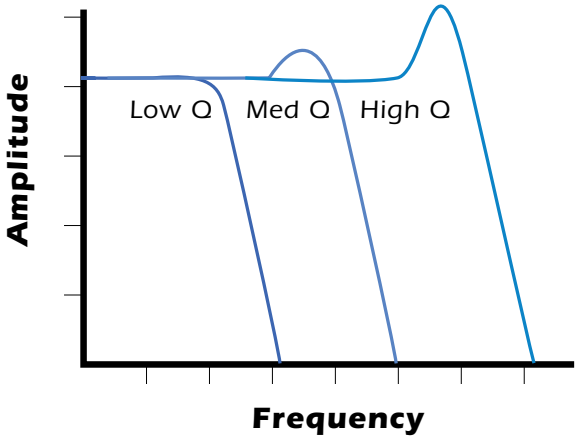


A filter that only lets a certain band of frequencies pass is called a band-pass filter.

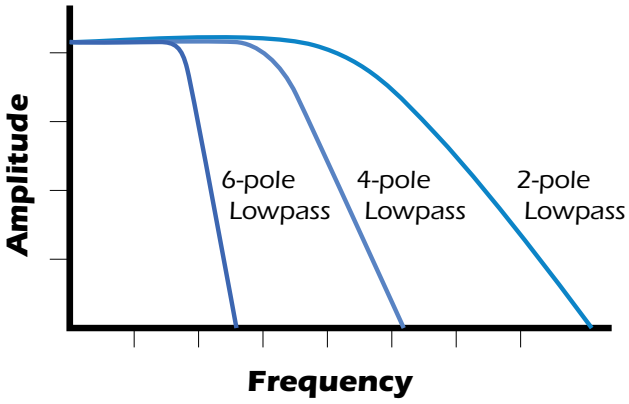


A Notch Filter is just the opposite of a bandpass filter and is used to eliminate a narrow band of frequencies.

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. The Proteus 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.



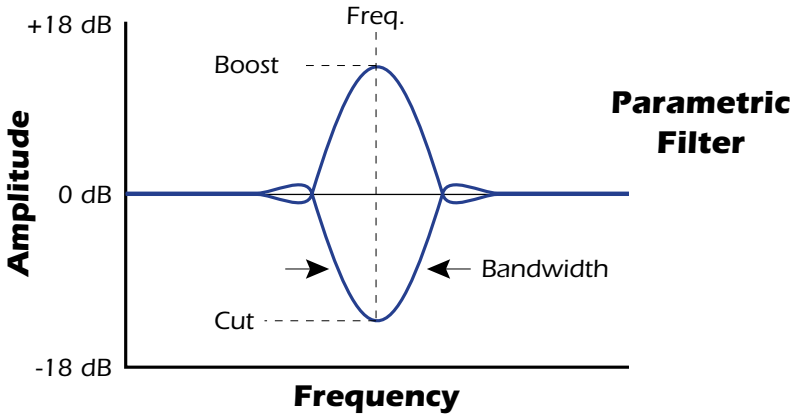
Using a filter, we now have a way to control the harmonic content of a sampled sound. As it turns out, even a simple low pass filter can simulate the response of many natural sounds.

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

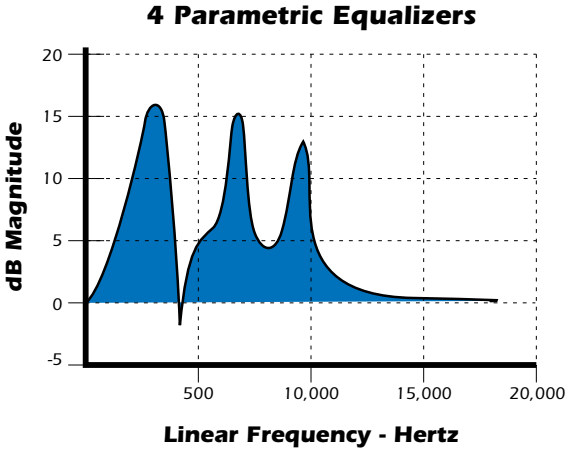
A more complex type of filter is called a parametric filter or Swept EQ. A parametric filter allows control over three basic parameters of the filter. The three parameters are: Frequency, Bandwidth, and Gain. The Frequency parameter allows you to select a range of frequencies to be boosted or cut, the 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.

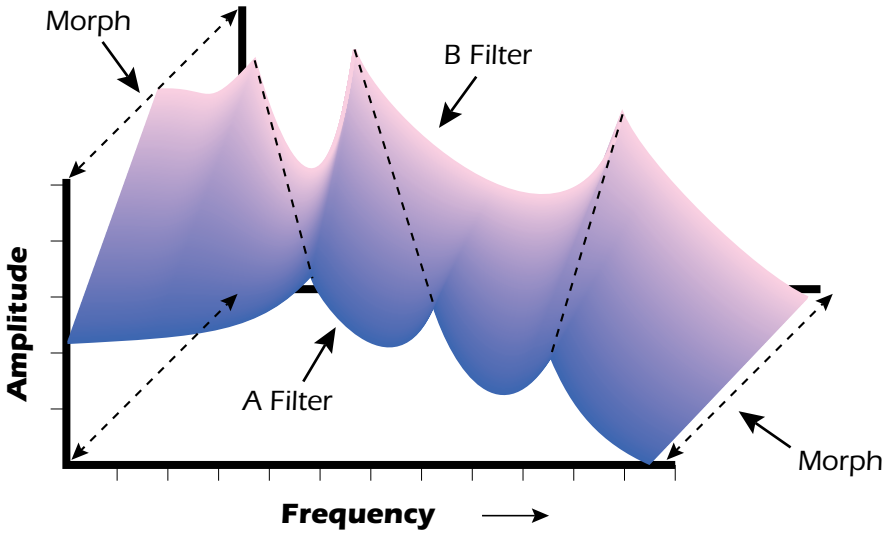
If four parametric filter sections were cascaded, it would be possible to create the following complex filter response.



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.

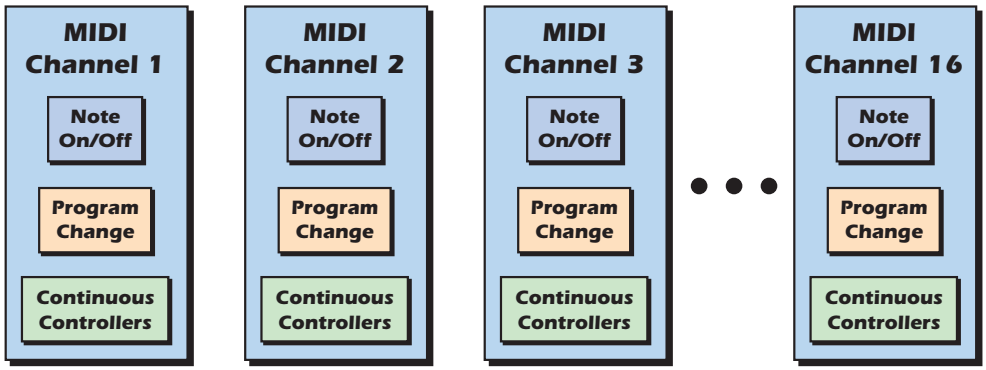
Consider, as an example, the human vocal tract, which is a type of complex filter or resonator. There are dozens of different muscles controlling the shape of the vocal tract. When speaking, however, we don't think of the muscles, we just remember how it feels to form the vowels. A vowel is really a 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 Proteus 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.

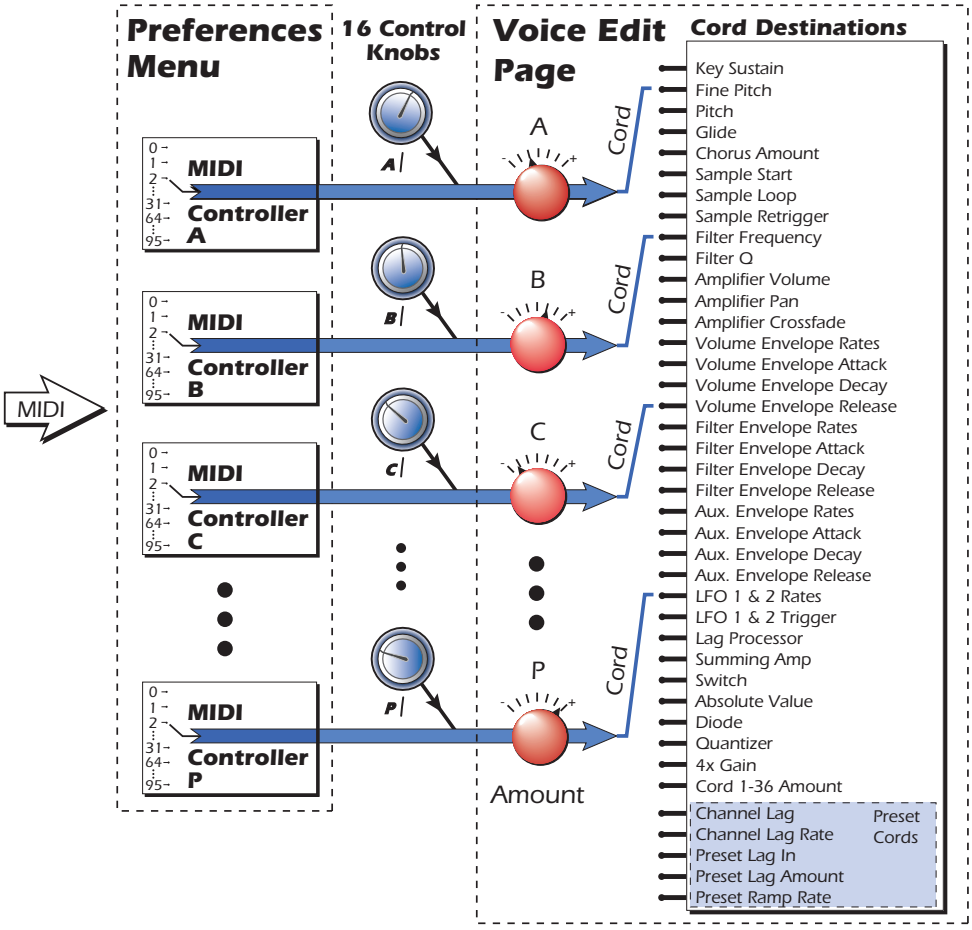


Any MIDI controller can be routed to any modulation destination. First, you have to know which controller numbers your keyboard transmits. Most modern MIDI keyboards let you select a controller number for each control on the keyboard. For example, it may let you select a number from 0-31 for the data slider. The realtime controller numbers that the keyboard transmits must match the numbers Proteus 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. Proteus 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 Editor page.

MIDI Program Change Commands

The Proteus X receives MIDI Program Change commands on each of its 32 MIDI channels. MIDI program changes select Proteus 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

The Proteus X accepts MIDI bank select commands using MIDI continuous controller numbers. Continuous Controller (CC) 0 is the MSB (most significant byte) and CC 32 is the LSB (least significant byte). Normally you send both the MSB and LSB controllers to implement a bank change.

Proteus 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.

❖ **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.

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.

Musicians wanted *more* presets and so 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).

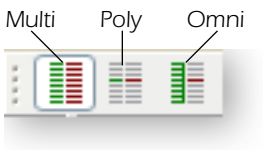
Because Bank Selects are implemented using Continuous Controllers, the Bank Selections can be made *per channel*. (This is getting better and better.) For each MIDI channel, you can select any of 16,384 banks and then one of the 128 presets in the bank. Of course no synthesizer has 16,384 banks (yet), but hey, it's nice to know it's possible (for that really BIG project).

MIDI Modes

There are three MIDI modes used by the Proteus 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. The Proteus X has the ability to play a different preset on each of 32 MIDI channels (using two MIDI ports of 16 channels each).

▪ The general term for "Preset" in MIDI-speak is "Program". Same thing.



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.

10 - Appendix

The Proteus X File Converter

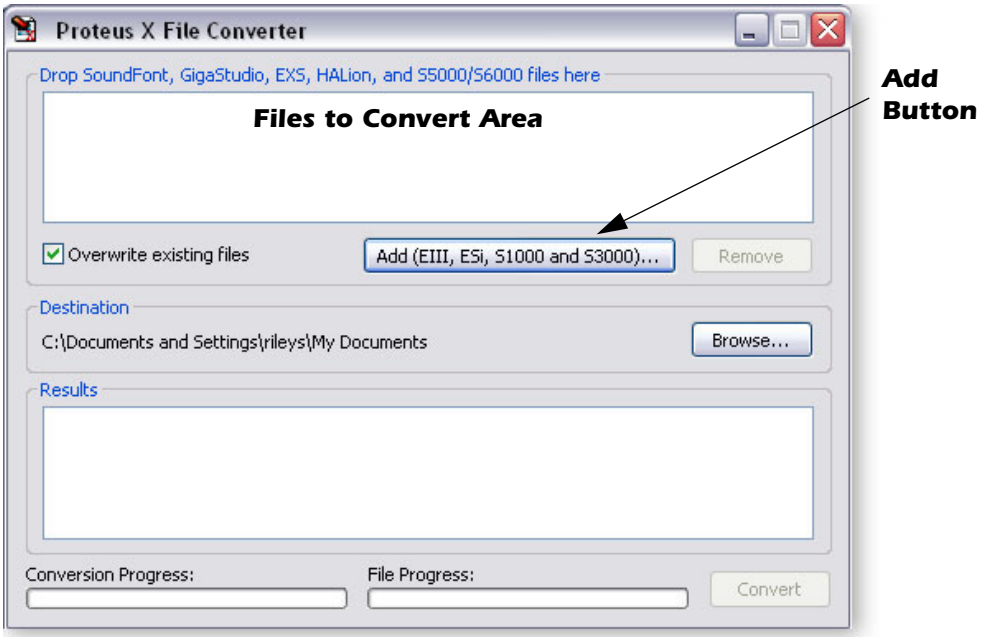
The Proteus X Converter is a special application that enables you to convert most popular sampler formats into the Proteus X's .exb file format, for use in your Proteus X. Any source media may be used, such as CD-ROM's, fixed hard drives, ZipDisks, all removable media, magneto-optical drives, and all others. Exception: floppy media is not supported, however, see notes on possible workarounds.

Supported File Types

The Proteus X file Converter can convert the following types of files into Proteus X format:

- Akai S-1000/3000
- Akai S-5000
- Emu E3/ESi
- Tascam GigaStudio
- SoundFont
- Emagic EXS-24 Mark I and Mark II
- Steinberg HALion I and II

If you don't have sounds for any of these products, you won't need to use the File Converter. Keep the application installed in case you ever want to convert file formats.



IMPORTANT: You don't have to convert Emulator IV banks!

Simply locate the desired EIV bank using the **System Tab**, then **Open** or **Merge** the bank.

▶ To Use the File Converter:

1. **SoundFont GigaStudio, EXS, HALion, and S5000 files:**
   Drag and drop files from your hard drive to the **Files to Convert** area.

   EIII, ESi, S1000, and S3000 files:
   Press the "Add Button" below the **Files to Convert** area, then select the files from your CD-ROM or hard disk.

2. Select a **Destination** location (Browse).
3. Press **Convert**. The file Converter does the rest, preserving loop, sample and preset information.
4. Files may be removed from the **Files to Convert** area by first selecting them and then clicking the **Remove** button.

- **Overwrite Existing Files** - Overwrites files with the same name in the **Destination** folder.

▼ *E4/E3 Library CDs use a proprietary format and can only be read using SCSI/ATAPI drives.*

Background

The Proteus X Converter is the result of over eight years of painstaking conversion work, and piles of research on sampler behavior and performance. The conversions into the Proteus X will be of extremely high quality - so good that sometimes you won't even notice that you are using the sounds in a different sampler!

The Proteus X Converter is able to convert single files, multiple files, and entire disks - and even more than one disk - simultaneously. Simply click the Add button, and a customized file browsing dialog will appear.

From this dialog, simply choose disks, folders, or files you want to convert. You can select single files by double-clicking on them, or by clicking on the dialogs Add button. You can select multiple-selected files, or folders/disks, by selecting them and clicking on the dialog's Add button.

Remember that selecting a folder to translate includes not only all the files that are immediately within that folder, but all files within all the folders within that folder. In other words, it always recurses every relevant folder tree.

The objects you want to convert are added to the Converter Application's object list. They are not converted yet. You may add or delete from the list until you're ready to convert.

Also note that if you choose a folder of, say, EXS-24 files, they are entered into the Converter Application list individually. You can see the name of the file and the path that it exists in.

Source Format Information

The Proteus X Converter is able to browse any sort of disk, whether it is PC-formatted, Mac-formatted, or specially formatted for a specific sampler.

Below is helpful background on each source format, and some relevant details on how their special features are exported into the Proteus X.

Akai S-1000/3000

The Akai S-1000 was one of the first hardware samplers using the 16-bit format. The S-3000 came out a couple years afterward, offering expanded programming and expanded structures.

The Akai format is proprietary; you cannot view the contents of the disk using the standard file browsing mechanisms on a computer. There are files that can be written to

DOS that represent the Akai format; they use the .ak1 or the .ak3 file extension. The Proteus X Converter supports these.

The Akai format is made of these elements:

Partitions: Maximum 60mb in size, one or more of them on a drive.

Volumes: Exist inside a Partition, these hold the Programs and Samples.

An Akai Volume is the equivalent of a Proteus X Bank. The Proteus X Converter can convert whole disks, Partitions, or Volumes. You cannot convert single Programs, which are roughly the equivalent of an Emu Preset.

The Akai S-1000/3000 can stack Programs; in other words, play two Programs at the same time, by setting each Program to the same "Program Number" and the same MIDI Channel in their parameters. The Proteus X Converter will automatically combine these into the same Proteus X Preset, to mimic exactly how the Akai treats these Programs.

The Akai S-1000/3000 use two separate samples to perform stereo operations, usually marked "-L" and "-R" in their names. The Proteus X Converter will generally combine samples that were meant to be stereoized; please see "Combining Dual-Mono Samples into Stereo Files" on page 167 for more specific information.

The Proteus X Converter expects that the samples files referenced by the Akai Programs within the Volumes will be in the same folder as the Program, just like the Akai itself behaves. The vast majority of the time they are. See "Locating Referenced Samples" on page 166 for more information.

Akai S-5000

The Akai S-5000 uses files stored on regular computer format disks, and uses files with the .akp extension for their Programs, and these reference .wav files to store their samples. These can be mono or stereo.

Each .akp file is the equivalent of a Proteus X Bank, although when you are converting whole folders, each folder will convert to a single Proteus X Bank, with each .akp file becoming a Preset within that Bank.

Akai .akp files and the .wav files they are connected to may be quite large, up to 512mb.

The Proteus X Converter expects that the .wav files referenced by the .akp will be in the same folder as the .akp, just like the Akai itself behaves. See "Locating Referenced Samples" on page 166 for more information.

Emu E3/ESi

The Emu E3 Series by now is the most legendary sampler ever produced. In fact it's so awesome that it's scary to even convert its format to anything else.

The Emu E3 Series format, which includes the Emulator 3, the Emulator 3x, the ESi-32, ESi-4000, and ESi-2000, is proprietary; you cannot view the contents of the disk using the standard file browsing mechanisms on a computer. There are files that can be written to DOS that represent the Emu E3 format; they use the .e3, .e3x, or the .esi file extension. The Proteus X Converter supports these.

The Proteus X structure is similar if not exact to the E3's; an E3 Bank is equivalent to a Proteus X Bank.

Tascam GigaStudio

The Nemesys Gigasampler was the first product that became GigaStudio. Its sampler software played endless waves (technology developed and licensed from Rockwell) directly from disk. Gigasampler does provide looping, and that is good. Gigasampler uses the DOS disk format, since it is a Windows computer program.

Nemesys then expanded their product line to include the Gigastudio, which is essentially Gigasampler 2.0; It has an improved interface, added functionality, and greater polyphony. Gigasampler and GigaStudio are very similar - we will refer to them as Giga.

The basic Instrument unit on Giga is the Instrument, or .gig file. A gig file can hold up to 128 Instruments; they hold the parameters and the wavedata to be played. An Instrument holds a maximum 4 (GigaStudio maximum 8) keymaps, arranged in a horizontal mapping system called Regions (same as a Keygroup on Akai samplers, except they do not overlap). You can have a Region for every key on the keyboard, and a Region can specify up to 4 (8 for GigaStudio) samples (called Layers). Stereo interleaved samples are supported (they take up two Layers, of course), but a Region cannot hold both Stereo and Mono samples. You can also specify up to 32 velocity splits (wow), but this is lessened the more Layers you use. The envelopes are semi-regular PADSR's.

Giga also supports Performance files (.prf in Giga 1.0, .gsp in Giga 2.0), which are essentially macros that load Instruments into the different MIDI channels. The Proteus X Converter supports the Giga 1.0 .prf format, as well as 2.0 .gsp GigaStudio files.

There are quite a few commercial Giga libraries that use a special compression algorithm; these files won't permit .wav extraction, and make the file smaller in size by about 20%. These .wav files aren't looped.

One very interesting thing about Gigasampler is that the file format had a copy-protection feature on it. A sound-development company would be able to produce files that cannot be loaded unless a code is entered; after that, the sound resides in that computer while being authorized by that code. This could have helped protect the company from having pirated samples floating around. Interestingly, GigaStudio does not retain this protection, although it gives you some hard time dialogs that post warnings.

SoundFont

Introduced in 1993, the SoundFont sample-based synthesis format has become a standard with the proliferation of the Creative Technology sound cards. SoundFont technology is a sample format that was invented by E-MU for the purpose of creating a flexible wavetable synthesis solution for Creative Labs. E-MU added their expertise and created a solution that would be embraced for consumer and professional applications.

SoundFonts are .wav file samples that have been transformed by a SoundFont editor, such as Vienna, into MIDI-controllable instruments. They're also referred to as .sf2 files, patches or programs, and are generally put together in groups known as SoundFont Banks, which can contain definitions of up to 128 instruments and one drum set.

Emagic EXS-24 Mark I and Mark II

The Emagic EXS-24 is the internal sampler that is available for Emagic's popular Logic DAW. It also comes in a VST flavor called the EXSP-24 that can be used within or outside of Logic.

EXS-24 files can exist on PC- or Mac-formatted disks. The Proteus X Converter can access and convert both types, as it can view both PC and Mac drives.

The Mark I and Mark II use the same file format; the Mark II just has more features. The EXS-24 uses the .exs file extension, although, especially in the case of Mac files, it is not

necessary to have the file extension. The Proteus X Converter will only see EXS-24 files on PC-formatted disks if they have the .exs extension; on Mac disks, they either can have the .exs extension or have the EXS-24 Creator code and file-type.

An EXS-24 file does not hold the samples; they are referenced and can be in the form of AIFF, WAVE, Sound Designer 1 or 2, or Mac System 7 files.

The EXS-24 is very popular; however, its programming ability is somewhat limited, which makes it fairly easy to convert into the Proteus X's superior framework. However, there are certain features that need special addressing in Proteus X. Please see "Keyswitching & Other Non-Proteus X Features" on page 166 for more information.

The EXS-24, like the Proteus X, uses streaming-like technology to load and play large sets of samples. The Proteus X Converter will convert any amount and size of samples, and the Proteus X is powerful enough to play any size.

An EXS-24 file is the equivalent of a Proteus X Bank, although when you are converting whole folders, each folder will convert to a single Proteus X Bank, with each EXS-24 file becoming a Preset within that Bank.

The EXS-24 files store the locations of its referenced samples within its files. The samples do not necessarily have to be in the same folder as the EXS-24. The Proteus X Converter expects that the sample files referenced by the EXS-24 file will be where it says they will be, just like the EXS-24 itself behaves. See "Locating Referenced Samples" on page 166 for more information.

Steinberg HALion I and II

The Steinberg HALion is a popular VST based software sampler. HALion files can possibly exist on PC- or Mac-formatted disks. The Proteus X Converter can access and convert both types, as it can view both PC and Mac drives.

Both HALion I and II use the same file format; II just has more features. HALion files use the .fxp file extension, although, especially in the case of Mac files, it is not necessary to have the file extension. The Proteus X Converter will only see HALion files on PC-formatted disks if they have the .fxp extension; on Mac disks, they either can have the .fxp extension or be file-typed with the HALion Creator code and file-type.

It is important to note that the .fxp extension is commonly used by other VST-related files. The Proteus X Converter only shows the .fxp files that are verified internally as being HALion-compatible ones.

A HALion file does not hold the samples; they are referenced and can be in the form of AIFF, WAVE, Sound Designer 1 or 2, or Mac System 7 files.

HALion has extensive programming capabilities, including a large modulation matrix. The Proteus X's larger CORD capabilities are more than enough to handle the load; however, there are certain features that need special addressing in Proteus X. Please see "Keyswitching & Other Non-Proteus X Features" on page 166 for more information.

HALion, like the Proteus X, uses streaming-like technology to load and play large sets of samples. The Proteus X Converter will convert any amount and size of samples, and the Proteus X is powerful enough to play any size.

A HALion file is the equivalent of a Proteus X Bank, although when you are converting whole folders, each folder will convert to a single Proteus X Bank, with each HALion file becoming a Preset within that Bank.

HALion files stores the locations of its referenced samples within its files; please see "Locating Referenced Samples" on page 166 for more information on how the Proteus X Converter handles these.

Notes...

Keyswitching & Other Non-Proteus X Features

GigaStudio, EXS-24, and HALion have some particular features that the Proteus X Converter attempts to recreate in some innovative ways. Below is a list of the major ones.

Key Switching: This is a newly-found feature where the user hits a non-sounding key on the music keyboard, and that tells the sampler to change sample references to point to (usually) a different set of samples. The Proteus X does not support this, so the Proteus X Converter splits each keyswitching section into separate Presets, followed by the phrase "KSW x" (where x is the split number).

Controller Switching: This is just like the above, only the references are changed upon receipt of a controller message. For example, the mod wheel can be set up to reference one sample in the down position, when it hits value 40 the sample reference changes to another sample, etc. The Proteus X Converter translates this by including all the samples and changing the amplitude where the sample comes in per the controller. This is not exactly what is done with the source file. The Proteus X is forced to use up polyphony whereas Giga/EXS/Halion actually turns the samples on and off, saving polyphony.

Release Triggering: This is when a sample is programmed to play when the engine receives a note-off message, rather than the normal note-on. This is not possible with the Proteus X, and such references are ignored.

Locating Referenced Samples

In any multisample format, such as the Proteus X's, there are two elements: the Program and the Sample.

The Program is defined as the information that determines what sample will be played when a certain MIDI note is triggering the engine. Programs are called many things on different platforms - on the Emu they are Presets, on the Ensoniq they are called Instruments, and so on.

The Sample is...well, the sample data and maybe a couple of immediate parameters, such as the loop points and the sample rate.

Some formats store all this information in one file; this is called the Monolithic Method. A good example of this is a GigaStudio .gig file. A .gig file can be huge if it contains long samples or a large quantity of samples.

Other formats store the Program information in a small file, sometimes called the control file. In that file, there are file references that instruct the sampler engine to load a sample from another file. These sample references can be broken by deleting or moving a sample file, or in certain instances, by moving the control file. This organizational method is called the Separate File For Samples Method. (Or, it should be called that.)

In this method, the built-in mechanism for finding the sample that is inherent in the Monolithic Method is gone. The Separate File For Samples Method needs a method to find where the separate file is in order to load it. Below are the ways this is accomplished by the supported formats.

Absolute Paths

Formats Who Use this Method: EXS-24, HALion

These control files store a complete absolute path to the sample. An absolute path looks something like this: c:\Program Files\My Kazoo Samples\hohner.wav.

Relative Paths

Formats Who Use this Method: None

These control files store relative paths, which look like this: ..\..\My Kazoo Samples\hohner.wav. This means "from the control file, go up two folders, then descend to this file".

Fixed Location Folder

Formats Who Use this Method: Akai S-5000, Emulator X, Proteus X

These formats simply assume that the sample is in the same folder as the control file, or, in the case of the Proteus X, they must be in a folder called SamplePool that is in the same folder as the control file.

Mac Catalog/FSSpec Reference

Formats Who Use this Method: HALion

Mac-oriented files, such as HALion files, use the Mac-format disk system to locate their sample files (this is called Catalog Searching, using a FSSpec, but never mind that for now).

Monolithic Files

Formats Who Use this Method: GigaStudio, Emu E3/E3x/ESi

These files store the samples within the Program files, thus the samples are always in the files and do not need "locating."

Special Notes

Akai S-1000/3000: This uses separate sample files within its own proprietary format, and it is possible for a Program to reference missing samples, but not usually.

When the Proteus X Converter encounters the Separate File For Samples Method, it uses the control file's reference to find the sample and expects it to be there. If it is not, the Proteus X Converter will still write the reference in the Proteus X's .exb file, pointing to an empty location so you may fill it in later manually within the Proteus X .exb file.

Combining Dual-Mono Samples into Stereo Files

Many formats do not support single Stereo files, so they simulate stereo by using two separate files (commonly they were sampled in simultaneously in true stereo imaging), layering them, and panning one hard left and the other hard right. Commonly they are named with the same sample/file name plus a "-L" and "-R".

The Proteus X Converter will automatically look for samples like this and combine them when possible. On the whole, every instance will be combined, but there are a couple reasons when they will not be. Below are the specific qualifications necessary when combining dual mono files into Proteus X stereo files.

Naming: The ends of the samples must have the same name including spaces, plus the endings of the sample names need to be similarly formatted with a "L" and "R". Examples: "-L" and "-R", "L" and "R", "(L)" and "(R)", etc.

Loop Points: The loops points need to be the same. If they are not, the Proteus X Converter will not combine the samples.

Sample Sizes: Most of the time, the Proteus X Converter WILL combine these, and make the new file the size of the largest sample.

Exact Alternative Parameters: Some formats store Root Key, Sample Rate, Fine Tune, and other informations with the sample. These need to be exactly the same.

Identical References: There are occasions where a Program, or several Programs that are slated to go into the same Bank, will reference both the samples at one point but only one of them in another. The Proteus X Converter will combine the two samples but also write a mono version to satisfy the mono reference.

Floppy Disk Support

The Proteus X Converter does not support proprietary floppy formats, such as Ensoniq, Akai, Roland, Emu, or other.

However, there is a way to get your sounds off the floppies using other programs onto your hard drive in a form in which the Proteus X Converter can convert them to Proteus X formats. You can use various DOS programs written by various people over the years. You can get more information on these at:

www.chickensys.com

Please be advised that these are frequently not supported and do not run on Windows XP, but please see the site for some easily attained workarounds.

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 Proteus X, will typically sound like white noise. Proteus 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".

Proteus X VSTi

Launching the VSTi Application

The following instructions apply to Cubase 5.1. Other host applications will have their own unique methods for launching a VSTi application. If you are using a host application other than Cubase 5.1, consult the documentation that came with your host application.

▶ To Launch the Proteus X VSTi in Cubase 5.1

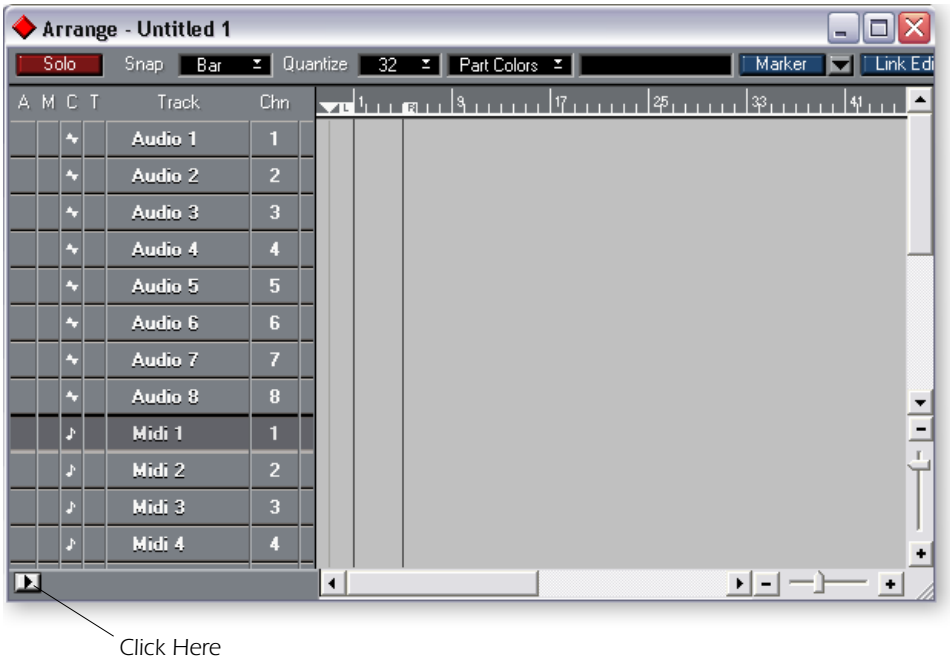
❖ **Tip:** When running multiple VSTis, distribute the CPU% among the VSTis. Example: If one VSTi does not glitch at 60%, set the CPU% of two VSTis so they add up to 60%.

PatchMix DSP

1. Click the **New Session** button on the toolbar.
2. Select **Default MultiTrack** session for this tutorial.

Cubase

3. Start Cubase 5.1 from the Start Menu or from the desktop icon. The following window appears.



4. Select one of the **MIDI Tracks** as shown above.
5. **Click the Inspector icon** shown above to open the **Track Info**.
6. Select "**Panels, VST Instruments**" from the menu bar. The VST Instruments window appears.
7. Click on "No VST Instrument" and select **ProteusXVSTi** from the list.
8. In the **Track Info** section, click on the "Instrument" field. Select **ProteusXVSTi**.
9. To the right of the Proteus X VSTi selection, a pop-up channel selection screen appears. Select **Any**.

Proteus X

10. In the VST Instruments window, click **Edit**. Proteus X VSTi will open.
11. From the File menu in Proteus X VSTi, select **Open** and select a bank to open.
12. Play your MIDI keyboard. The MIDI Activity lights should flash on Proteus X.

Checklist for Launching Proteus X VSTi

Cubase 5.1

Choose the following settings in the **Options** menu.

- **Options, Audio Setup, System**

  Audio I/O ..... **E-MU ASIO**

- **Options, MIDI Setup, System**

  MIDI Input ..... **E-DSP MIDI port (B800)**

- **Options, MIDI Setup, System**

  MIDI Thru..... **Enabled, Thru Off Channel- Off**

PatchMix DSP

Make sure you have an ASIO 1 & 2 input strip in the mixer.

VST Multisetup

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.

Keyboard Shortcuts

File Menu

New ..... Ctrl + N
Open ..... Ctrl + O
Merge ..... Ctrl + M
Save ..... Ctrl + S

Edit Menu

Undo ..... Ctrl + Z
Redo ..... Ctrl + Y
Cut ..... Ctrl + X
Copy ..... Ctrl + C
Paste ..... Ctrl + V
Select All ..... Ctrl + A

Options

All Sound Off ..... Ctrl + Pause

Window

Close ..... Alt + F4
Tile Horizontally ..... Ctrl + Shift + H
Tile Vertically ..... Ctrl + Shift + V

Presets

New Preset ..... Ctrl + W
Open Voices ..... Ctrl + Enter
New Voice ..... Ctrl + W
New Sample Zone ..... Alt + W

Links

New Link ..... Ctrl + W

View

Refresh F5

Voice Editor

(Place the cursor over the knob, slider or wheel)

Center Control (Pan, etc.) ..... Ctrl + Click
(Number key 5 also centers control when Num Lock is off)
Minimum Setting ..... Home
Maximum Setting ..... End

Right Click Shortcuts

Use the right mouse button to call up a popup window with these context-sensitive shortcuts.

On Instrument (Proteus X icon)
Open Bank..... Ctrl + O
Merge ..... Ctrl + M
Save As
Preferences

On Preset Icon (tree)
Open in New Window
Rename.....F2
Cut..... Ctrl + X
Copy ..... Ctrl + C
Paste ..... Ctrl + V
Delete ..... Del
Duplicate.....Ctrl + U
Category
Export
Select on Current Channel ..... Ctrl + P

On Link Icon (tree)
Open in New Window
New Link..... Ctrl + W

On Voice Processing Icon (tree)
Open in New Window

On a Sample Icon (tree)
Open in New Window
Rename.....F2
Cut..... Ctrl + X
Copy ..... Ctrl + C
Paste ..... Ctrl + P
Delete ..... Del
Duplicate.....Ctrl + U
Audition
Category

Voice Processing (Synth)
Template List
Save Template
Organize Templates

Voices & Zones
(Multisample name field)
Open Voices .....Ctrl + Enter
Open Voices in New Window
Cut Voice .....Ctrl + X
Copy Voice .....Ctrl + C
Paste Voice ..... Ctrl + P
Delete Voices
Duplicate Voices
Combine Voices
Expand Voice
New Sample Zone .....Alt + W
New Group
Select Group "N"

Voices & Zones
(Sample name field)
Open Sample.....Alt + Enter
Open Sample in New Window
Delete Sample Zones
Duplicate Sample Zones

Numeric Field
Undo
Cut
Copy
Paste
Delete
Select All

Toolbars
Checkboxes for Tools view: Status, Standard, Toolbars Dialog

Multisetup Icon (in tree)
Open in New Window
New .....Crtl + W

Drag & Drop

You can use simple "Drag & Drop" operations for many Proteus X functions.

Drag & Drop Operations

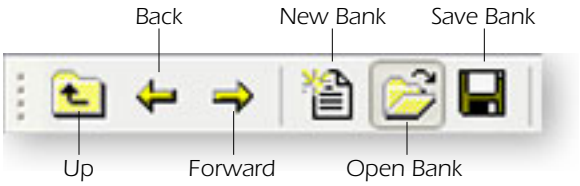
From	Drag & Drop To...	Action
Bank File in Windows	Proteus X Icon	Load or Merge Bank
Bank File in Windows	Proteus X System Folder	Copy Move Bank
Audio File in Windows	Instrument	Import
Audio File in Windows	Samples Folder	Import
Audio File in Windows	Folder in Windows	Copy/Move
Proteus X Bank File	Proteus X Icon	Load/Merge
Proteus X Bank File	Folder in Windows	Copy/Move
Proteus X Bank File	Proteus-X System Folder	Copy/Move
Proteus X Bank Preset	Proteus X Icon	Merge
Proteus X Bank Preset	Presets Folder	Merge
Proteus X Bank Preset	Multisetup Slot	Merge & place
Proteus X Bank Preset	Folder in Windows	Export as Bank
Proteus X Bank Preset	Proteus-X System Folder	Export as Bank
Proteus X Bank Preset	Link	Merge & Replace Link
Proteus X Bank Preset	Link (null space)	Merge & Add Link
Preset	Multisetup Preset Slot	Place
Preset	Folder in Windows	Export as Bank
Preset	Proteus-X System Folder	Export as Bank
Preset	Link	Replace Link
Preset	Link (null space)	Add Link
Multisetup	Instrument	Load
Instrument	Multisetup Folder	Save
Instrument	Multisetup	Replace
E4 Bank File	Instrument	Load/Merge
E4 Bank File	Folder in Windows	Copy/Move
E4 Bank File	Proteus-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

Drag & Drop Operations

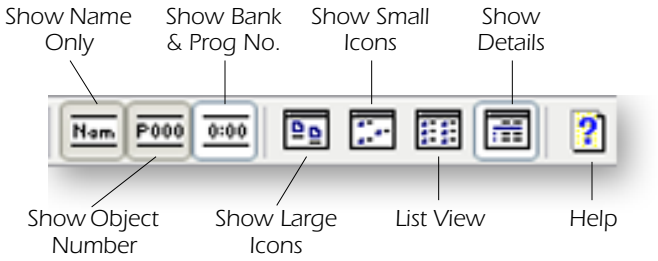
From	Drag & Drop To...	Action
E4 Bank Preset	Proteus-X System Folder	Export as Bank
E4 Bank Preset	Link	Merge & Replace Link
E4 Bank Preset	Link (null space)	Merge & Add Link
E4 Bank Sample	Instrument	Merge
E4 Bank Sample	Samples Folder	Merge
E4 Bank Sample	Folder in Windows	Export as Bank/Wave/AIFF
E4 Bank Sample	Proteus-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)	Folder in Windows	Copy/Move
E4 Bank File (CD)	Proteus-X System Folder	Copy/Move
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)	Folder in Windows	Export
E4 Bank Preset (CD)	Proteus-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)	Folder in Windows	Export as Bank/Wave/AIFF
E4 Bank Sample (CD)	Proteus-X System Folder	Export as Bank/Wave/AIFF
E4 Bank Multisetup	Instrument	Merge
E4 Bank Multisetup	Multisetup Folder	Merge

Toolbar Functions

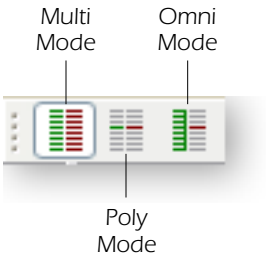
Standard Tools



Display Items

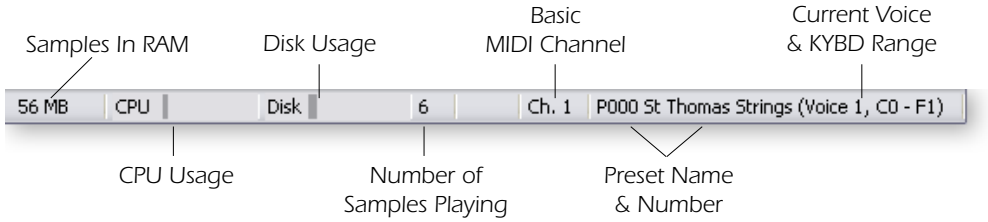


MIDI Modes



▼ E4/E3 Library CDs use a proprietary format and can only be read using SCSI/ATAPI drives.

Status Bar



MIDI Implementation Chart

MIDI Information	Transmitted	Recognized	Remarks
MIDI Channels	No	1-16, 17-32	32 MIDI Channels
Note Numbers	No	0-127	
Program Change	No	0-127	
Bank Select Response?	No	Yes	MSB + LSB
Modes: Omni *(Mode 1)* Mono *(Mode 2)* Poly *(Mode 3)* Mode 4 *(Y/N)* Multi *(Mode 5)*	No No No No No	No No No No No	Responds to all MIDI channels
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 Sample Dump Standard File Dump MIDI Tuning Master Volume Master Balance Notation Information Turn GM1 System On Turn GM2 System On Turn GM1 System Off Other *(See Remarks)*	No No No No No No No No No No No	No No No No No No No No No No No	
NRPNs	No	No	
RPN 00 *(Pitch Bend Sensi.)* RPN 01 *(Chan. Fine Tune)* RPN 02 *(Chan Coar. Tune)* RPN 03 *(Tuning Prog Sel.)* RPN 04 *(Tuning Bank Sel.)* RPN 05 *(Mod Depth Rang)*	No No No No No No	No No No No No No	
MIDI Timing & Sync	No	No	
MIDI Clock	Yes	Yes	
Song Position Pointer	No	No	
Song Select	No	No	
Start Continue Stop	No No No	No No No	
MIDI Time Code	No	No	
MIDI Machine Control	No	No	
MIDI Show Control	No	No	
General MIDI Compat? Is GM default mode?	No No	No No	
DLS compatible? Import DLS Files? Export DLS Files?	No No No	No No No	
Import Std MIDI files Export Std MIDI files	No No	No No	

MIDI Implementation Chart

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			*
38	Data Entry LSB			*
39	Chan Volume LSB			*
40	Balance LSB			*
41				*
42	Pan LSB			*
43	Expression LSB			*
44	Effect Cntl 1 LSB			*

Control #	Function	Transmitted	Recognized	Remarks
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 Cntrlr 6 ➳	Yes	Yes	*
76	Sound Cntrlr 7			*
77	Sound Cntrlr 8 ➳	Yes	Yes	*
78	Sound Cntrlr 9 ➳	Yes	Yes	*
79	Sound Cntrlr 10	Yes	Yes	*
80	Gen Purp Cntrlr 5	Yes	Yes	*
81	Gen Purp Cntrlr 6			*
82	Gen Pur Ctrlr 7 ➳	Yes	Yes	*
83	Gen Pur Ctrlr 8 ➳	Yes	Yes	*
84	Portamento Cntrl			*
85				*
86				*
87				*
88				*
89				*
90				*

Control #	Function	Transmitted	Recognized	Remarks
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				
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.

NOTES: * Proteus X can receive ANY continuous controller number from 1 to 95. Because of Proteus 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 Proteus X.

✸ Value of 0 = reset all except vol & pan; value of 127 = reset all.

Other: Pan: -64 = hard left, +63 = hard right

(part 2 - Controllers)

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 vv	
Note On	9n kk vv	velocity 0 = note off
Key Aftertouch	An kk vv	kk = 0-127 vv = 0-127
Program Change	Cn vv	0-127
Channel Aftertouch	Dn vv	0-127
Pitch Bend	En ll mm	l = lsb, m = msb
Real-time Controller	Bn cc vv	cc = 00-31, 64-95
Footswitch	Bn cc vv	cc = 64-79, vv ≥ 64 = on
Volume	Bn 07 vv	0-127
Pan	Bn 0A vv	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

Index

Symbols

Numerics

A

B

C

D

E

F

G

H

I

J

K

L

M

N

O

P

Q

R

S

T

U

V

W

Z