E-MU

1212m   1820   1820m

Digital Audio System

Owner's Manual

CREATIVE
PROFESSIONAL

E-MU Digital Audio System

Owner's Manual

Software Version: 1.81

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1- Introduction

Welcome!

Thank you for purchasing the E-MU 1820M, E-MU 1820 or E-MU 1212M digital audio system. Your computer is about to be transformed into a powerful audio processing workstation. We've designed this E-MU digital audio system to be logical, intuitive and above all, to provide you with pristine sound quality. All three systems offer unprecedented value by providing studio-quality, 24-bit/192kHz multi-channel recording and playback at an astounding price.

E-MU Digital Audio System Components

E-MU 1212M	E-MU 1820	E-MU 1820M
▪ E-MU 1010 PCI Card ▪ E-MU 0202 I/O Daughter Card ▪ 0202 I/O Card Cable ▪ (2) MIDI Adapter Cables ▪ **D.A.S. Software/Driver Installation CD-ROM** ▪ **Prod. Tools Software Bundle CD-ROM** ▪ Quick Start Guide	▪ E-MU 1010 PCI Card ▪ AudioDock ▪ EDI (E-MU Digital Interface Cable) ▪ PC Power Adapter Cable ▪ Headphone Splitter Cable ▪ **D.A.S. Software/Driver Installation CD-ROM** ▪ **Prod. Tools Software Bundle CD-ROM** ▪ Quick Start Guide	▪ E-MU 1010 PCI Card ▪ AudioDockM ▪ E-MU Sync Daughter Card ▪ Sync Card Cable ▪ EDI (E-MU Digital Interface Cable) ▪ PC Power Adapter Cable ▪ Headphone Splitter Cable ▪ **D.A.S. Software/Driver Installation CD** ▪ **Prod. Tools Software Bundle CD** ▪ Quick Start Guide
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All Systems Include:

The E-MU 1010 PCI Card is the heart of all three systems. Its powerful hardware DSP processor allows you to use over 16 simultaneous hardware-based effects, which place minimal load on your computer's CPU. The Firewire port provides high-speed connectivity to the Creative NOMAD® portable digital audio player, external CD-RW drives and other Firewire compatible devices such as DV camcorders, printers, scanners and digital still cameras. The E-MU 1010 PCI Card also provides eight-channels of ADAT® optical digital input and output, as well as a S/PDIF stereo digital input and output.

The PatchMix DSP mixer application is included in all the systems. PatchMix DSP delivers unmatched flexibility in routing your audio between physical inputs and

outputs, virtual (ASIO/WAVE) inputs and outputs and internal hardware effects and buses—no external mixer needed. You can add digital effects, EQs, meters, level controls and ASIO/WAVE sends anywhere you like in the signal chain.

Because the effects and mixing are hardware-based, there is no latency when you record. You can even record a dry signal while monitoring yourself with effects! Mixer setups can be saved and instantly recalled for specific purposes such as recording, mixdown, jamming, special effect setups, playing games, watching DVDs, or general computer use.

E-MU 1212M System

The E-MU 1212M includes the 0202 Daughter Card, which provides 2 line level, balanced analog inputs, 2 line level, balanced analog outputs, plus MIDI input and output. This is no-compromise audio interface, using ultra-high performance 24-bit/192kHz A/D - D/A converters to deliver an unbelievable 120dB dynamic range.

E-MU 1820 System

The E-MU 1820 includes the AudioDock, which is a half rack-space, audio interface. The AudioDock adds the following input and output capabilities to the system: two mic/line inputs with TFPro™ preamps, 6 balanced line level analog inputs, an RIAA stereo turntable preamp, 8 balanced line level outputs, an assignable headphone output, two sets of MIDI I/O ports, an additional S/PDIF optical output, and four stereo mini phone jacks for easy connection to powered speaker systems. Combined with the digital I/O on the 1010 PCI card, you have a total of 18 inputs and 20 outputs! Of course, professional-quality, 24-bit A/D and D/A converters with automatic DC blocking are used throughout.

E-MU 1820M System

The E-MU 1820M system includes the AudioDockM, and is a no compromise, mastering-grade system, which includes all the features of the 1820 system. The 1820M system is distinguished by the addition of ultra-high performance 24-bit/192kHz A/D - D/A converters which deliver an unbelievable 120dB dynamic range.

The Sync Daughter Card comes standard with the 1820M system and can be purchased as an optional upgrade to the 1820 and 1212M systems. The Sync Card adds Word Clock in and out for sample-synchronizing outboard digital equipment and SMPTE longitudinal time code in/out for syncing other recording equipment. A separate MIDI Time Code output port on the Sync Card eliminates timing problems caused by combining MTC with MIDI performance data.

You'll want to keep up with the latest software and options for your E-MU digital audio system. You can find all of this, plus other helpful information, at the E-MU Website: http://www.emu.com.

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.*

2 - Installation

Setting Up the Digital Audio System

There are six basic steps to installing your E-MU system:

1. Remove any other sound cards you have in your computer. (Once you are sure that the E-MU card works properly, your old sound card can be reinstalled if desired.)

2. Install the E-MU 1010 PCI card in your computer. Go there.

3. Install the 0202 Daughter Card or Sync Daughter Card (if applicable). Go there.

4. Connect the AudioDock (if applicable).

5. Install the PatchMix DSP software onto your computer.

6. Connect audio, MIDI and synchronization cables between the E-MU system and your other gear.

Notes for Installation

- IF AT ANY TIME DURING THIS INSTALLATION YOU SEE NO RESPONSE: Use the Alt-Tab feature to select other applications. One of them may be the Microsoft Digital Signature warning. It is possible for this warning to appear behind the installation screen.

- Make sure you have the latest Windows Service Packs from Microsoft (Windows 2000 - SP 4, Windows XP - SP 1 or higher).

- Disable onboard sound and uninstall all other sound cards. (If you wish to try using multiple sound cards in your system, do so after you have confirmed that your E-MU Digital Audio System is operating normally.)

- InstallShield "IKernel Application Error" on Windows XP: When installing this software on Windows XP, you may be confronted with a "kernel error" at the very end of installation. This is an issue with the InstallShield program, which is what we use to install software on your computer. Please do not be alarmed by this, as the error is innocuous.

  To read more about this error, and obtain instructions on how to avoid getting the message, please visit this website: http://support.installshield.com/kb/view.asp?articleid=q108020

- Multiple Digital Audio System sound cards are not supported.

Please read the following sections as they apply to your system as you install the E-MU 1010, paying special attention to the various warnings they include.

Prior to installing the hardware, take a few moments to write down the 18-digit serial number, which is located on the back of the box and on the 1010 PCI Card. This number can help EMU Customer Service troubleshoot any problems you may encounter—by writing the number down now, you'll avoid having to open your computer to find it later on.

Safety First!

- To avoid possible permanent damage to your hardware, make sure that all connections are made with the host computer's power off. **Unplug the computer's power cable to make sure that the computer is not in sleep mode**.

- Take care to avoid static damage to any components of your system. Internal computer surfaces, the E-MU 1010 PCI board and the interfaces are susceptible to electrostatic discharge, commonly known as "static." Electrostatic discharge can damage or destroy electronic devices. Here are some procedures you can follow when handling electronic devices in order to minimize the possibility of causing electrostatic damage:

- Avoid any unnecessary movement, such as scuffing your feet when handling electronic devices, since most movement can generate additional charges of static electricity.

- Minimize the handling of the PCI card. Keep it in its static-free package until needed. Transport or store the board only in its protective package.

- When handling a PCI card, avoid touching its connector pins. Try to handle the board by its edges only.

- Before installing a PCI card into your computer, you should be grounded. Use a ground strap to discharge any static electric charge built up on your body. The ground strap attaches to your wrist and any unpainted metal surface within your computer. If you don't have a ground strap, you can ground yourself by touching the metal case of another piece of grounded equipment.

- Before connecting a cable to your interface or between PCI cards, touch the connector sleeve of the cable to the sleeve of the jack to which you'll be connecting the cable in order to discharge any static build-up.

Connector Types

These connector types are used to connect the E-MU 1010 hardware components. They will be referred to by the name shown in the first column of the following chart:

Name	Description	Connects
Card/External	CAT5 Connector	1010 PCI card and AudioDock
S/PDIF In	RCA Connector	S/PDIF digital audio devices
S/PDIF Out	RCA Connector	S/PDIF digital audio devices
ADAT Optical In	TOSLINK Optical Connector	ADAT digital audio devices (or S/PDIF)
ADAT Optical Out	TOSLINK Optical Connector	ADAT digital audio devices (or S/PDIF)
1394	Firewire Connector	Interfaces to Firewire peripherals

▼ **Warning:** The E-MU 1010 PCI Card has been designed to use readily available and inexpensive standard computer system cables to make it easy for you to find replacement cables if your original cables become damaged or lost. However, because these standard cables types are used for other purposes, you must use caution to avoid connecting the cables incorrectly. Please verify that all cables are connected only to the proper components before powering up your system.

Installing the E-MU 1010 PCI Card

Note: This installation is very simple but if you are not familiar with the installation of computer peripherals and add-in boards, please contact your authorized E-MU Systems dealer or an approved computer service center to arrange for the installation.

▶ **To install the 1010 PCI card into your computer**

1. Make sure that the power switch on your computer is off.
   IMPORTANT: Unplug the power cord from the wall outlet!

2. Touch a metal plate on your computer to ground yourself and to discharge any static electricity.

3. Follow the computer manufacturer's recommended procedure for opening the case.

4. Remove the metal brackets from two adjacent PCI slots. If you have the E-MU 1820 system (non-M) you only need to remove the bracket from a single PCI slot. Put the screw(s) aside for use later. See figure 1 below.

▪▪ **Note:** Some computer cases don't use screws to secure PCI cards. In this case, follow the instructions that came with your computer.

Figure 1

PCI Slots

ISA Slots
(may not be present on your computer)
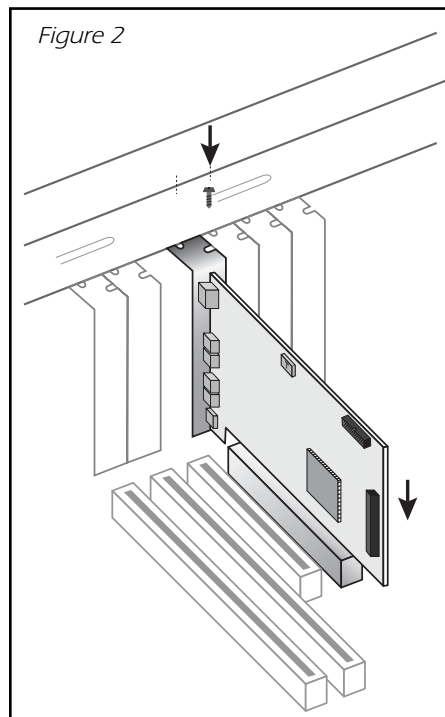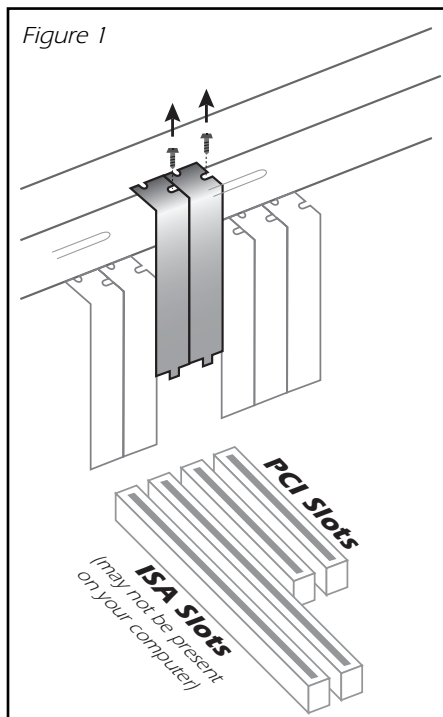
Figure 2

5. Align the E-MU 1010 PCI card with the slot and press gently but firmly down into the slot as shown in figure 2.

6. Do not force the E-MU 1010 card into the slot. Make sure that the gold finger connector of the card is aligned with the PCI bus connector on the motherboard before you insert the card into the PCI slot. If it doesn't fit properly, gently remove it and try again.

7. Secure the card into the slot using one of the screws you placed aside earlier.

Install the Sync Daughter Card or 0202 Daughter Card

- **E-MU 1820M** - If you're planning to use Word Clock. MIDI Time Code or SMPTE sync, unwrap the Sync Daughter Card and get ready to install it. If you don't need these options or don't have an empty PCI slot, you can skip these next few steps.

- **E-MU 0202M** - Unwrap the 0202 Daughter Card and get ready to install it.

1. Connect the ribbon cable provided with the kit between the E-MU 1010 card and the 0202 Daughter card or Sync Daughter Card as shown in figures 3 & 4. The cables are keyed so they cannot be incorrectly inserted. Seat the connectors firmly in the sockets and arrange the cables neatly.

2. Align the Sync Daughter Card or the 0202 Daughter Card with the slot and press gently but firmly down into the slot as shown in figure 2 on the preceding page.

3. Do not force the E-MU Card into the slot. Make sure that the tab at the rear of the card is aligned with the PCI bus connector on the motherboard before you insert the card into the PCI slot. If it doesn't fit properly, gently remove it and try again.

4. Secure the card into the slot using one of the screws you placed aside earlier.

Figure 3

Sync Daughter Card

Figure 4

0202 Daughter Card
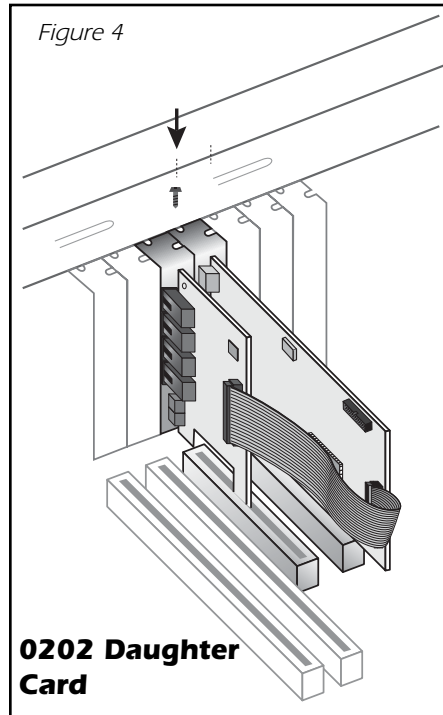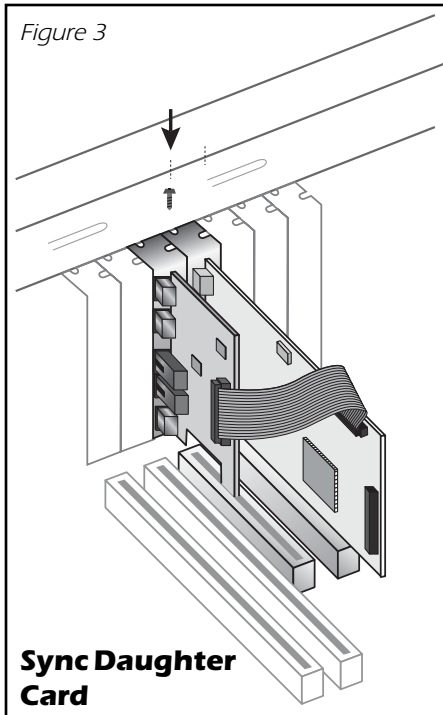
E-MU 0202 & AudioDock

If you have both the E-MU 0202 I/O card and the AudioDock, you are advised not to connect both to the same E-MU 1010 PCI card using this version of software. There are known issues with doing this what will be addressed in a future software update.

AudioDock Owners only

5. Locate the Disk Drive Power Converter Cable shown below and identify the large male connector (the one with pins on it). Plug this connector into a spare disk drive power cable in your computer. If there is no spare disk drive power cable, insert the Adapter Cable between one of your disk drives and the power supply.

▼ *The AudioDock requires 1.1 Amps at 12V (13 Watts) to operate. The AudioDockM requires 1.25 Amps at 12V (15 Watts) to operate.*

Power Converter Cable

To Disk Drive

To Power Supply
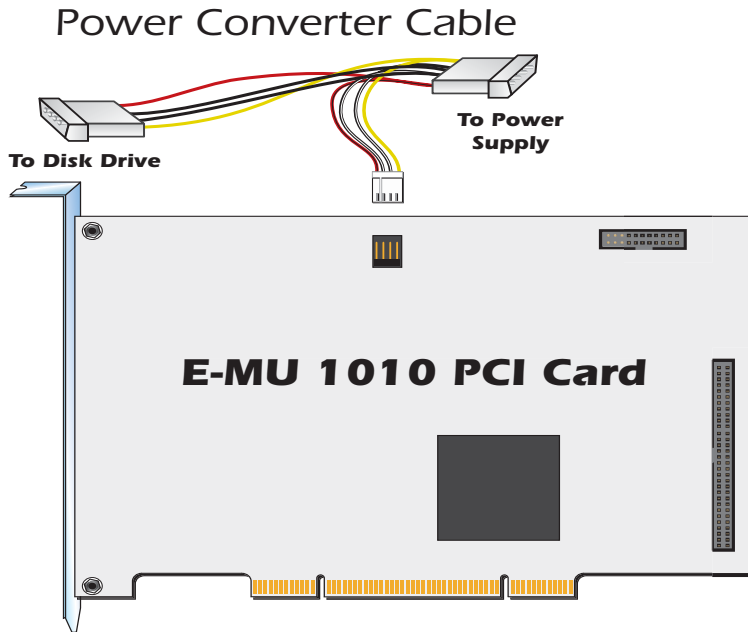
E-MU 1010 PCI Card

6. Plug the small connector into the E-MU 1010 PCI card as shown above. The connector is keyed and can only be inserted one way.

7. After all components have been installed and securely fastened, close the computer case.

8. Connect the supplied network-type cable from the 10 BaseT jack on the E-MU 1010 PCI card labeled "EXTERNAL" to the matching connector labeled "Card" on the AudioDock. The cable supplied with the AudioDock is specially shielded to prevent unwanted RF emissions.

9. Plug the power cord back into the wall outlet and turn on your computer.

▼ **CAUTION:** *Do not connect the supplied CAT5 cable to the Ethernet or network connector on your computer. Doing so may result in permanent damage to either your computer, the E-MU 1010 or both.*

Rubber Feet

Four rubber feet were included with the AudioDock. These feet should be used if you're not going to rack mount the AudioDock. If you are going to rack mount the AudioDock, leave the feet off.

To install the rubber feet, simply peel off the protective backing from the adhesive and press the feet into the round depressions at each corner of the bottom plate.

Rubber Foot   **Rubber Foot**

Rack-shelf mount   **Rack-shelf mount**

Rack-shelf mount   **Rack-shelf mount**

Rubber Foot   **Rubber Foot**
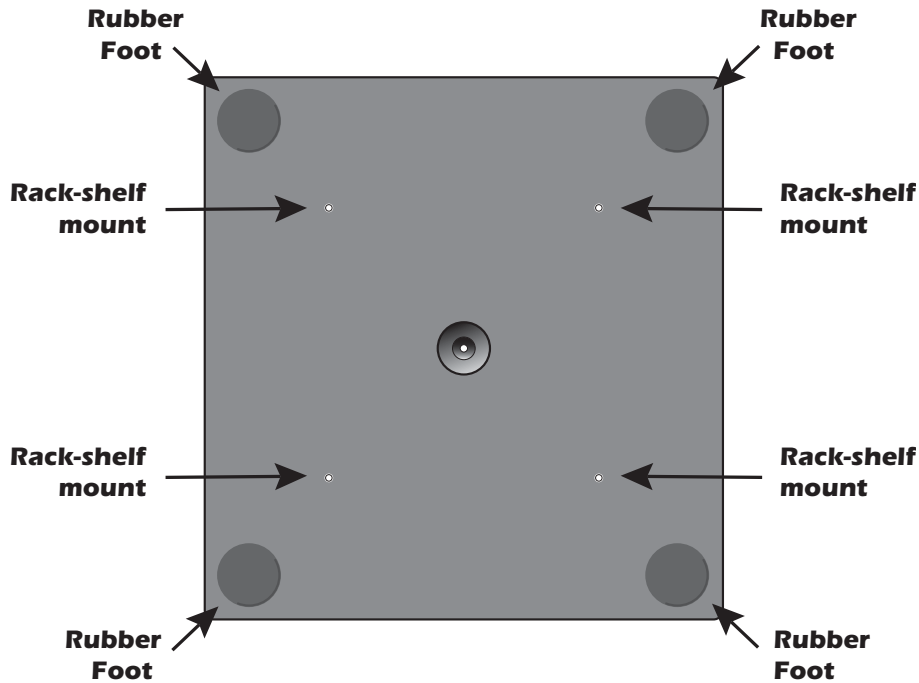
Rack Mounting the AudioDock

The AudioDock was designed to be rack mounted using standard 19-inch rack shelves. (*These shelves are available from a number of sources on the Internet. Search for "Rack Shelf".*) Two AudioDocks fit side by side on a single rack shelf. Two screws are provided to secure the AudioDock to the rack shelf (M3 x 6mm). Do not use screws longer than 6mm or damage to the circuit board may result.

Software Installation

Installing the E-MU 1010 Drivers
The first time you restart your PC after installing the E-MU 1010 PCI card, you will need to install the PatchMix DSP software and E-MU 1010 PCI card drivers.

Windows 2000 or Windows XP
The software is not compatible with other versions of Windows.

1. After you have installed your Digital Audio System, turn on your computer. Windows automatically detects the Digital Audio System and searches for device drivers.

2. When prompted for the audio drivers, click the **Cancel** button.

3. Insert the E-MU software Installation CD into your CD-ROM drive. If Windows AutoPlay mode is enabled for your CD-ROM drive, the CD starts running automatically. If not, from your Windows desktop, click **Start->Run** and type **d:\setup.exe** (replace **d:** with the drive letter of your CD-ROM drive). You can also open the CD and double-click **Setup.exe**.

4. The installation splash screen appears. Follow the instructions on the screen to complete the installation.

5. Choose **"Continue Anyway"** when you encounter the "Windows Logo Testing" warning screen. See the note below for more information.

6. When prompted, restart your computer.

> ☷ **Serial Number -**
> *During the registration process, you will be asked to enter your 18-digit serial number. The serial number is located on the back of the box and on the 1010 PCI Card.*

Uninstalling all Audio Drivers and Applications
At times you may need to uninstall or reinstall some or all of the audio card's applications and device drivers to correct problems, change configurations, or upgrade outdated drivers or applications. Before you begin, close all audio card applications. Applications still running during the uninstallation will not be removed.

1. Click Start -> Settings -> **Control Panel**.

2. Double-click the **Add/Remove Programs** icon.

3. Click the **Install/Uninstall** tab (or Change or Remove Programs button).

4. Select the E-MU driver/application entries and then click the **Add/Remove** (or Change/Remove) button.

5. In the InstallShield Wizard dialog box, select the **Remove** option.

6. Click the **Yes** button. Restart your computer when prompted.

7. You may now re-install existing or updated E-MU 1010 PCI card device drivers or applications.

Note About Windows Logo Testing
When you install the Digital Audio System drivers, you will see a dialog box that informs you that the driver has not passed Windows Logo testing.

The Digital Audio System drivers are not signed because the driver does not support some of the consumer audio features that the Microsoft driver signing program requires, most notably Digital Rights Management.

However, the Digital Audio System drivers have been rigorously tested using the same test procedures that a signed driver requires, and it passes in all important categories, including those that measure the relative stability of the driver. So, it is perfectly safe to install these drivers on your computer.

3 - PCI Card & Interfaces

The E-MU 1010 PCI Card

The E-MU 1010 PCI card is the heart of the system and contains E-MU's powerful E-DSP chip. The powerful hardware DSP on this card leaves more power free on your CPU for additional software plug-ins and other tasks.

Connections

Connects to Audio Dock via EDI Cable

S/PDIF In/Out

ADAT or S/PDIF Optical In/Out

Firewire
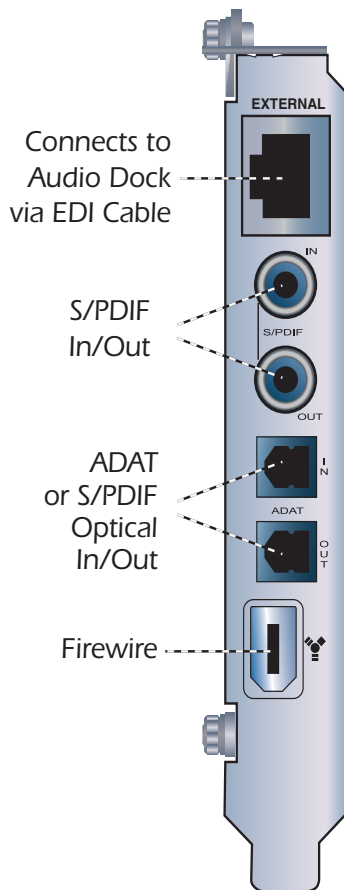
EDI Connector
Connects to the AudioDock using the supplied EDI cable. This cable provides a a two-way data link between the E-MU 1010 and the AudioDock as well as supplying power to the AudioDock.

S/PDIF Digital Audio Input & Output
RCA phono jacks are standard connectors used for S/PDIF (Sony/Philips Digital InterFace) connections. Each jack carries two channels of digital audio. The E-MU 1010 receives digital audio data with word lengths of up to 24-bits. Data is always transmitted at 24-bits.

S/PDIF digital I/O can be used for the reception and/or transmission of digital data from external digital devices such as a DAT external analog-to-digital converter or an external signal processor equipped with digital inputs and outputs.

The S/PDIF out can be configured in either Professional or Consumer mode in the Session Settings menu. The 1010 PCI card can also send and receive AES/EBU digital audio through the use of a cable adapter. See AES/EBU to S/PDIF Cable Adapter for details.

The S/PDIF input and outputs are usable at the 44.1kHz, 48kHz 88.2kHz and 96kHz sample rates, but are disabled for 176.4kHz and 192kHz. The word clock contained in the input data stream can be used as a word clock source. See System Settings.

ADAT Optical Digital Input & Output

The ADAT optical connectors transmit and receive 8 channels of 24-bit audio using the ADAT type 1 & 2 formats. The word clock contained in the input data stream can be used as a word clock source. See System Settings. Optical connections have certain advantages such as immunity to electrical interference and ground loops. Make sure to use high quality glass fiber light pipes for connections longer than 1.5 meters.

At the 96kHz or 192kHz sample rates, the industry standard S/MUX interleaving scheme is used for ADAT input and output. S/MUX uses additional ADAT channels to achieve the required bandwidth. See the chart below or go here for additional information.

▼ **Important:** When using any type of digital I/O such as S/PDIF or ADAT, you MUST sample sync the two devices or clicks and pops in the audio will result.

Sample Rate	Number of Audio Channels
44kHz/48kHz	8 channels of 24-bit audio
88.2kHz/96kHz	4 channels of 24-bit audio, using S/MUX standard
176.4kHz/192kHz	2 channels of 24-bit audio, using S/MUX standard

IEEE1394 Firewire

This port allows high speed data transfer between your computer and external storage devices such as hard disks, CD-ROM drives, etc. Firewire ports are "hot-swappable" which means that you can plug and unplug Firewire peripherals without turning off power.

This port does NOT support Firewire audio. It is fully compliant with the OHCI 1.1 specification, supporting asynchronous and isochronous data transfers at 100, 200 or 400 Mbit/s with multiple DMA channels.

▼ **Important:** The 6-pin Firewire connector/port has a 3-watt maximum power output. Connect only one high power usage device such as a IEEE 1394 hard disk or CD-RW drive to this port unless it is self-powered.

The 0202 Daughter Card

The 0202 Daughter card is the companion card for E-MU 1010 systems which don't include the AudioDock. The 0202 Daughter card provides one pair of 24-bit balanced analog inputs and one pair of 24-bit balanced analog outputs, plus MIDI in and out.

Connections

Left / Right Line Inputs

Left / Right Line Outputs

MIDI In/Out
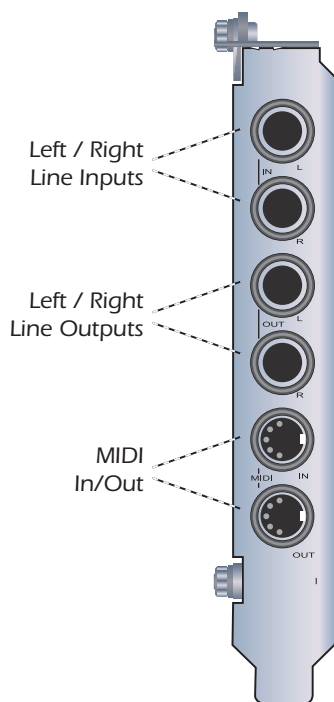
Analog Inputs and Outputs

The 0202 Daughter Card provides two balanced, analog inputs and two balanced, line level analog outputs. The inputs can be connected to any line level stereo signal from keyboards, CD-players, cassette decks, etc. The analog inputs are assigned to a mixer strip in the mixer application.

The outputs can feed any line level input such as a mixing board, the auxiliary input on your stereo or a set of powered speakers. The line outputs are not designed to drive headphones directly. Connect the line outputs to a stereo receiver or mixer with a headphone jack to obtain the proper current drive.

Either TRS (tip-ring-sleeve) balanced or TS unbalanced cables can be used. Balanced cables provide better noise immunity and +6dB higher signal level. The output line level can be set to accommodate the consumer -10dBV standard, or the pro audio +4 dBu standard in the I/O screen of the Session Settings dialog box. See I/O Settings.

MIDI In/Out

The MIDI input and output port can be assigned in your specific MIDI application. Connect the MIDI adapter cable that came with your 0202 Daughter card to the mini-DIN connectors on the card. The adapter cables convert the mini-DIN to standard DIN connectors used on most keyboards and synthesizers. Connect MIDI Out to the MIDI In port of your synthesizer and MIDI Out of your synth to MIDI In of the 0202 Daughter Card.

The AudioDock

The AudioDock connects to the E-MU 1010 PCI card via the EDI cable.

The AudioDock provides (6) balanced analog inputs, a pair of microphone preamp inputs, (8) balanced line-level analog outputs, (4) 1/8" outputs for connecting powered computer speakers, (2) MIDI inputs, (2) MIDI outputs, one optical S/PDIF output, a headphone amp, and a RIAA equalized turntable preamp section which is "normalled" into line input 3L and 3R.

The inputs are configured as follows:

(2)	mono microphone/line inputs
(3)	stereo pairs of line level inputs (6 inputs)
(1)	RIAA equalized turntable preamp input allows you to connect a turntable without using an expensive external preamp. *Note: These inputs are automatically disconnected when plugs are inserted into inputs 3L & 3R since the A/D converters are shared between the turntable inputs.*
(2)	MIDI input ports

The outputs are configured as:

(4)	stereo pairs of line level outputs
(1)	stereo pair driving a stereo headphone jack (the provided cable allows 2 stereo outputs)
(1)	optical S/PDIF output (stereo).
(4)	stereo 1/8" computer speaker outputs. These outputs carry the same signals as the 4 stereo line level outputs and are provided as a convenience for connecting computer speaker systems.
(2)	MIDI output ports

❖ The AudioDock is completely "hot pluggable"— It's OK to plug or unplug the AudioDock while the computer is turned on.

▼ It's a good idea to mute AudioDock inputs 3 in the PatchMix DSP mixer when nothing is plugged in, since the turntable preamp has a very high gain (60dB) and could contribute extra noise to your mix/ monitor bus.

Front Panel Connections
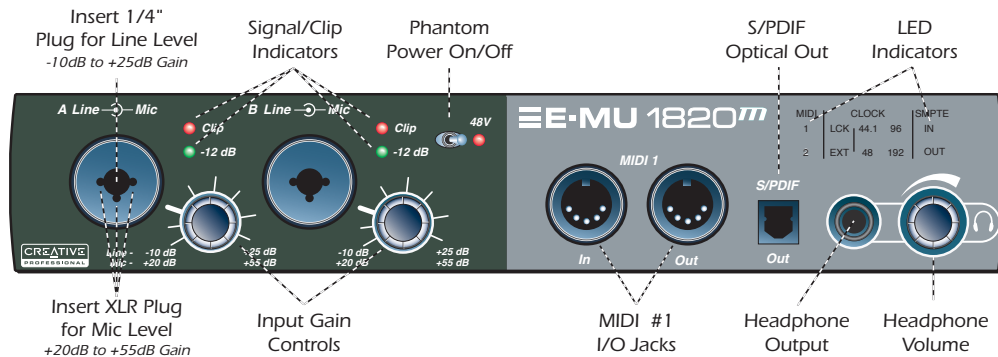
Preamp Section

The front panel mono Mic/Line inputs A & B can be used as balanced microphone inputs, hi-Z guitar pickup inputs, or line level inputs. The Neutrik combination jack accepts microphones using a standard XLR connector or line level/hi-Z inputs using a 1/4 inch TRS/TS connector.

The superb-sounding mic preamps are designed by TF Pro®. Each preamp has a level control which sets the preamp gain from +20dB to +55dB for the XLR input and from -10dB to +25dB for the line input. The line markings around the knobs are calibrated in 10dB increments. The heavy hash marks on the gain controls indicate unity analog gain to the converter inputs (~5dBV input = 0dBFS output).

A phantom power switch enables +48 volt phantom power supplied to both microphones. A red LED illuminates to indicate phantom power is enabled. See Phantom Power for additional information.

Each microphone input has its own input level and clipping indicators. The green LED indicates presence of signal and illuminates at -12 dB below clipping. The red LED indicates that the signal is clipping the input. These LEDs monitor the signal directly at the analog-to-digital converters and before any processing by the rest of the system. When setting the levels for signals being sent into the AudioDock, the red level indicators should never flash.

MIDI 1 In/Out

MIDI input and output ports allow you to interface any type of MIDI equipment such as keyboards, effect units, drum or guitar controllers. The MIDI drivers were installed when you installed your PatchMix DSP software and the MIDI ports will appear in your system control panel under "Sounds and Audio Devices".

S/PDIF Optical Out

The front panel S/PDIF connector is an optical TOSLINK output which, by default, carries a digital copy of the main output pair. This output is a convenient way to master to a portable DAT, MD recorder or other media. This S/PDIF output can also be freely assigned in the mixer application.

Headphone Output & Volume Control

The headphone output drives standard stereo headphones and the adjacent volume control sets the listening level. The headphone amplifier can drive headphones with impedance as low as 24 ohms. The headphone output uses a high-current version of the high-quality output amplifiers used on the other channels. For this reason it has a very clean signal that can be used as another stereo output if you need it. This output is freely assignable in the mixer application.

▼ **Warning:** Some microphones cannot tolerate phantom power and may be damaged. Check the microphone's specifications and requirements before using phantom power.

▼ **Warning #2:** After turning phantom power off, wait two full minutes before recording to allow the DC bias to drain.

The audio mutes for a second when phantom power is turned on.

❖ **Tip:** Since the headphone output can be placed into any insert location, you can use it to monitor or troubleshoot the signal flow.

The AudioDock Front Panel Indicators

```
MIDI        CLOCK        SMPTE
  1     LCK   44.1   96    IN
  2     EXT    48   192    OUT
```
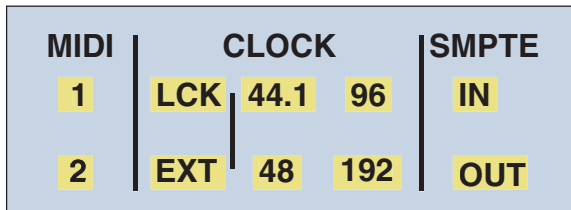
The MIDI Input Indicators
These two indicators, labelled 1 & 2, show MIDI activity on the MIDI input jacks.

The Clock Source and Sample Rate Indicators
These LED indicators on the front panel of the AudioDock show the current timing synchronization and sample rate. These indicators reflect the current settings in the Session Settings Window. See System Settings.

- If the sample rate is 88.2 kHz, the 44.1k and 96k LEDs both illuminate. If the sample rate is 176.4 kHz, the 48k and 192k LEDs both illuminate.

The Clock Source LEDs
The Clock Source LEDs indicate the source of the master clock that is currently driving the E-MU 1010.

LED	Clock Source
LCK	**Lock** - Indicates that the internal or external clock is locked and valid.
EXT	**External** - Indicates that an external clock source is selected.

When the system is running from an external or digital clock source, the AudioDock continually checks that the incoming clock source is valid. If the clock source changes or becomes invalid in any way, the LCK LED will flicker or will not be lit. If sync has been lost, the audio outputs will also be muted. The AudioDock will switch to internal clock at 48kHz if sync is lost and switch back to external clock if sync is re-established.

Typical causes of loss of digital or external sync include:

- Removing the S/PDIF or external clock cables

- Loss of power to the device providing the clock source

- Sudden changes in the S/PDIF sample rate
  (as would happen if a DAT tape had data recorded at multiple rates)

The Sample Rate Indicators
The Sample Rate LEDs indicate the current sample rate at which the system is running. The LEDs will light solidly to indicate the different sample rates of 44.1kHz, 48kHz, 96kHz or 192kHz.

When slaving to an external master source, the clock may drift slightly or change dramatically (i.e. abrupt sample rate change or unplugging of physical master source). the E-MU 1010 is tolerant to minor drifting within the supported rates of 44.1k, 48k, 88k, 96k, 176k and 192k, but if the sample rate drifts out of range (1%) the "Lock" LED will be extinguished. If set to external clock and the external clock is removed or out of tolerance, the E-MU 1010 will switch to internal clock at 48kHz (the default sample rate) until an external clock source is connected.

Rear Panel Connections

6 Balanced Line Level Inputs
(configured as 3 stereo pairs)

Turntable Inputs
(tied to line input 3)

Turntable Ground

Alternate Outputs
(same as outputs 1-4)

6 Balanced Line Level Outputs
(configured as 3 stereo pairs)

Monitor Outputs

MIDI Port 2 In/Out

Connect to E-MU 1010 Card
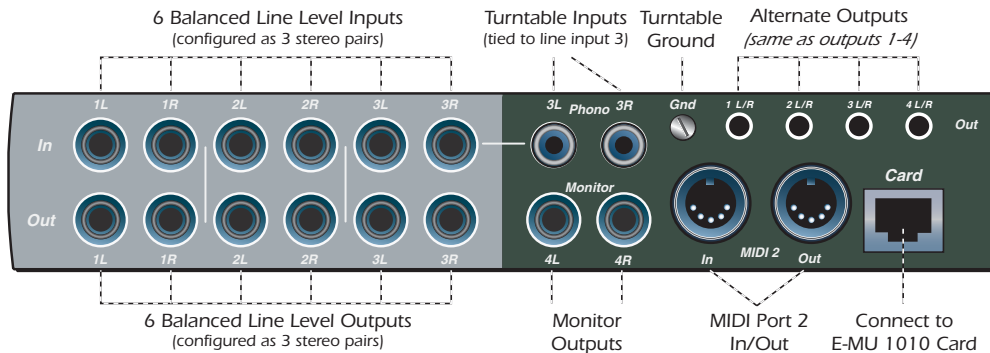
Line Level Analog Inputs

Six balanced 24-bit, line-level, analog inputs are provided (1-3). These can be used to input any line level signal from keyboards, CD-players, cassette decks, etc. The analog inputs are assigned to mixer strips in the mixer application. Input line level can be set to accommodate the consumer -10dBV standard, or the pro audio +4 dBu standard in the I/O screen of the Session Settings dialog box. See I/O Settings.

The maximum input level in pro mode is 18dBV (=20.2dBu). Maximum output level in consumer mode is 6dBV.

Either TRS balanced or TS unbalanced cables can be used. See page 101 for additional information about unbalanced cables and connectors. The line-level inputs are all servo-balanced, enabling them to convert unbalanced signals to balanced signals internally to reduce noise.

Turntable Inputs & Ground Lug

The RCA turntable inputs feed an RIAA equalized preamp designed to accept moving magnet type phono cartridges. The turntable inputs share line level inputs 3L and 3R. Inserting a plug into Line Input 3 disconnects the turntable preamp from that channel. Connect the ground lead from your turntable to the ground lug to prevent hum.

Line Level Analog Outputs

Eight balanced 24-bit, line-level, analog outputs are provided (1-4). Output pair 4 is designated as the Monitor Output and is fed by the monitor bus of the PatchMix DSP mixer application. We suggest that you plug your speakers in here. All the analog outputs can be freely assigned in the mixer application. Special anti-pop circuitry mutes the analog outputs when power is turned on or off.

Like the analog line inputs, either TRS balanced or TS unbalanced cables can be used. Balanced cables provide better noise immunity and +6dB higher signal level. The output line level can be set to accommodate the consumer -10dBV standard, or the pro audio +4 dBu standard in the I/O screen of the Session Settings dialog box. See I/O Settings.

The maximum input and output line levels are matched when the input and output settings are set to the same mode (pro or consumer) in the I/O preferences screen.

IMPORTANT NOTE:

Do NOT use balanced audio cables (TRS) when connecting balanced outputs to unbalanced inputs. Doing so can increase noise levels and introduce hum.

▼ **Warning:** Do NOT leave your turntable connected when using outputs 3L and 3R. This can cause a ground loop.

▪ It's also a good idea to mute Dock inputs 3 in the PatchMix DSP mixer when nothing is plugged in, since the turntable preamp has a very high gain (60dB) and could contribute extra noise to your mix/monitor bus.

Computer Speaker Analog Outputs

These stereo mini-phone (3.5mm) jacks duplicate line level outputs 1-4 with a lower output level to accommodate consumer speakers. These line level outputs are designed to interface easily with powered speakers.

Computer Speaker Output	Duplicates Line Level Output
1 L/R	Tip = 1L Ring = 1R
2 L/R	Tip = 2L Ring = 2R
3 L/R	Tip = 3L Ring = 3R
4 L/R	Tip = 4L Ring = 4R

MIDI 2 In/Out

A second, independent set of MIDI input and output ports which can be assigned in your specific MIDI application.

EDI Connector (Card)

Connects to the AudioDock to the E-MU 1010 PCI card using a CAT5 computer cable. The cable supplied with the AudioDock is specially shielded to prevent unwanted RF emissions.

▼ **Warning:** The AudioDock has been designed to use readily available and inexpensive standard computer system cables. This makes it easy for you to find replacement cables if your original cable becomes damaged or lost. However, because these standard cables types are used for other purposes, you must use caution to avoid connecting the cables incorrectly. DO NOT connect the supplied EDI cable to the Ethernet or network connector on your computer. Doing so may result in permanent damage to either your computer, the E-MU 1010 card, or the AudioDock.
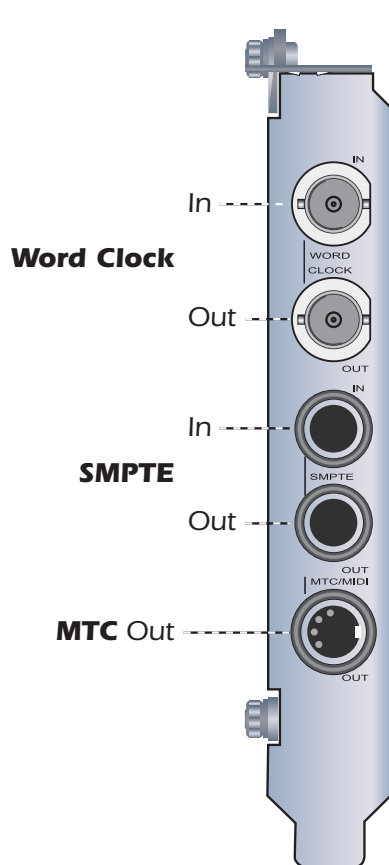
The Sync Daughter Card

The Sync Daughter card (included in the E-MU 1820M system and available as an option for other systems) provides word clock in and out, SMPTE (LTC) in and out and an additional MIDI output for transmitting MIDI Time Code (MTC). MIDI Time Code is a special rendering of SMPTE that can be transmitted over MIDI cables. For additional information about SMPTE, please refer to SMPTE Background.

Synchronization is a basic technique needed for connecting multiple pieces of equipment. Word Clock, S/PDIF or ADAT optical are all industry standard methods of synchronizing digital equipment together at the system sample rate (44.1kHz, 48kHz, 96kHz, or 192kHz). The master clock source is set in the Mixer Session Settings menu. See System Settings for more information.

Recording equipment can also be synchronized so that two audio recorders or an audio and video recorder can lock together as a single machine. SMPTE and MTC sync are used because they convey absolute time information. Word clock, S/PDIF or ADAT optical sync only synchronize the sample rate and unlike SMPTE and MTC, do not convey song position information. In a synchronized system, there is usually one MASTER machine, and one or more SLAVES. When the master starts, the others will follow (chase).

The Sync Daughter card is also a format converter. It converts incoming SMPTE time code to MIDI Time Code (MTC) and passes this information to the host computer to be used by a sequencer or audio recorder application. When your computer application is the "Master", the Sync Daughter card converts MTC into SMPTE and sends it out to another SMPTE device.

Connections

The Sync Daughter Card contains Word Clock inputs and outputs for clock signals used in a studio where a common sample rate reference is required to keep multiple pieces of digital equipment running together. This is referred to as "house clock" or "house sync," and is set to the actual sample rate of the system. Use a cable with BNC connectors to connect incoming clock signals to the Word Clock In jack on the Sync Daughter card. Connect the Word Clock Out to your other digital equipment to use the Sync Daughter Card as the Master Word Clock source. See Word Clock In/Out.

The Sync Daughter Card provides SMPTE (LTC) sync in and out, on two 1/4″ phone jacks. LTC can be recorded onto an unused audio track on an analog or digital recorder and then fed back into the SMPTE input to synchronize your computer sequencer/recorder. See SMPTE Background for more information.

MIDI Time Code is also output whenever MTC is being generated by the host application (sequencer or audio recorder). A special cable converts the mini DIN to a standard MIDI jack.

See the Sync Daughter Card Supplement for more information about using the Sync Daughter Card.

4 - The PatchMix DSP Mixer

PatchMix DSP

The PatchMix DSP Mixer is a virtual console which performs all of the functions of a typical hardware mixer and a multi-point patch bay. With PatchMix, you may not even need a hardware mixer. PatchMix DSP performs many audio operations such as ASIO/WAVE routing, volume control, stereo panning, equalization, effect processing, effect send/return routing, main mix and monitor control and allows you to store and recall these "Sessions" at will.

▶ **To Invoke the PatchMix DSP Mixer**

1. Left-click once on the E-MU icon ≡ on the Windows System Tray. The PatchMix DSP mixer window appears.

❖ *Click on the buttons and knobs in the mixer screen below to jump to the description of the control.*

Overview of the Mixer
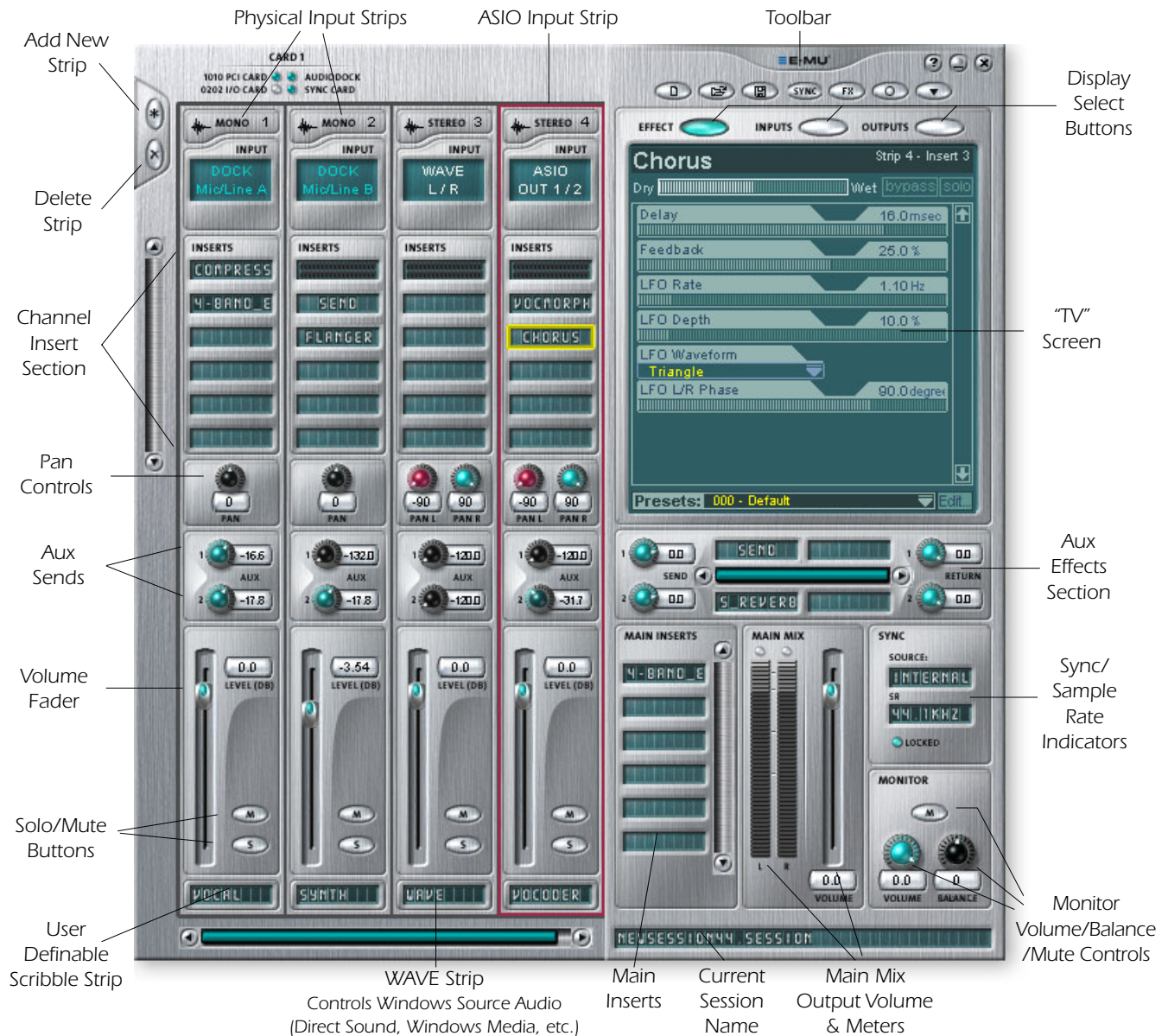
Mixer Window

The Mixer consists of four main sections.

Application Toolbar	Lets you manage sessions and show/hide the various views.
Main Section	Controls all the main levels, aux buses, and their inserts. This section also has a "TV" which shows parameters for the currently selected effect and the input/output patchbay. It also shows the session's current sample rate and whether it's set to internal or external clock.
Mixer Strips	This section is located to the left of the Main Section and shows all the currently instantiated mixer strips. Mixer strips can represent **Physical** analog/digital inputs, or **Host** inputs such as ASIO or Direct Sound. Mixer strips can be added or deleted as necessary. This section can be resized by dragging the left edge of the frame.
Effects Palette	This popup window is invoked by pressing the FX button in the toolbar. Iconic representations of all effects presets are shown here, organized by category. From this window, you can drag and drop effect presets into the insert slots available on the mixer strips and main section aux buses and main inserts.

A simplified diagram of the mixer is shown below.
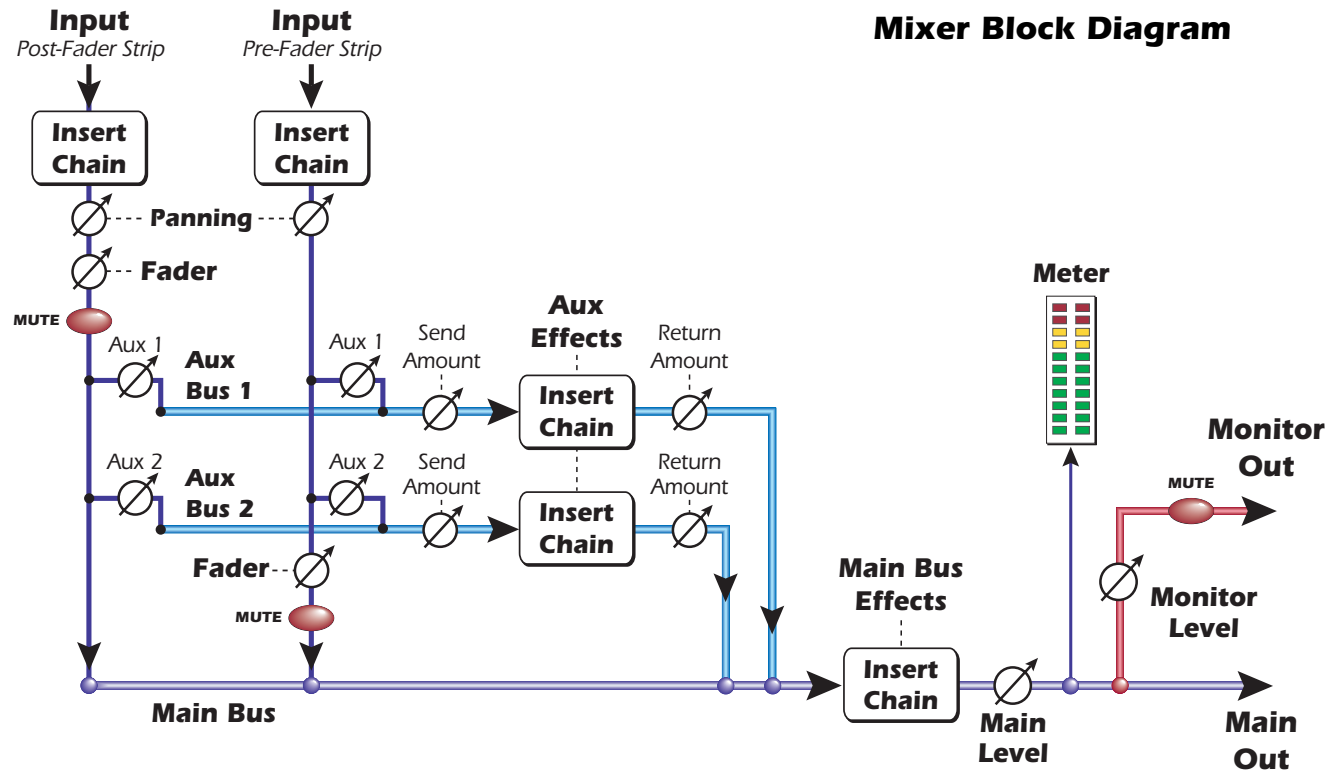
Pre Fader or Post Fader

When creating a new Mixer Strip, you have the option for the Aux Sends to be **Post** Fader (both Aux Sends come after the channel fader) or **Pre Fader** (both Aux Sends come before the channel fader). The Pre-fader option allows you to use either Aux Send as another mix bus, which is unaffected by the channel fader. More Information.

E-MU Icon in the Windows Taskbar

Right-clicking on the E-MU icon in the Windows taskbar calls the following window.

Right-Click Here ➝ ☰🕐 📋 11:28 AM

Edit... — Opens the PatchMix DSP Mixer.

Help... — Calls the PatchMix DSP help system.

Disable Splash Screen — Disables the splash screen that appears at boot-up.

Load FX on Startup ☑ — When unchecked, FX are not loaded until needed, resulting in faster computer boot.

Restore Defaults — Restores the default PatchMix DSP and driver settings.

Exit PatchMix DSP Services — Closes the PatchMix DSP background program, disabling use of all audio I/O from the E-MU hardware. Open the Patch-Mix DSP application to start audio again.

❖ **Restore Defaults**: Always try this option first if PatchMix is crashing or if you are having any other strange audio problems.

The Toolbar

New Session   Save Session   "About" PatchMix DSP   Session Settings

Open Session   Sync Settings   Show/Hide Effects   Global Prefs
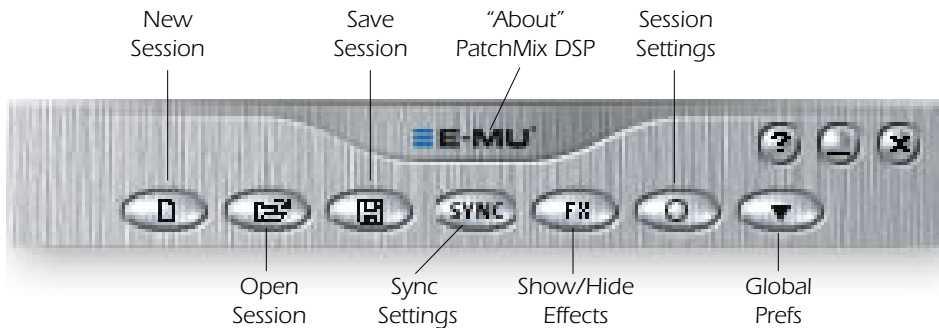
❖ Click the buttons in the toolbar to learn about their function.

New Session	Calls up the "New Session" dialog box. New Session.
Open Session	Calls up the standard "Open" dialog box, allowing you to open a saved Session.
Save Session	Calls up the standard "Save" or "Save As…" dialog boxes, allowing you to save the current Session.
Show/Hide Effects	Toggle button that shows or hides the FX palette.
Session Settings	Calls up the Sessions Settings window. Session Settings.
Global Preferences	Calls up the Global Preferences window.
Sync Settings	Calls up the SMPTE window. (if Sync Card is installed)
About PatchMix DSP	Right-Click on the E-MU logo to view the "About PatchMix DSP" screen, which provides the software and firmware version numbers and other information.

The Session

The current state of the PatchMix DSP mixer (fader settings, effects routings…everything!) can be saved as a Session. Whenever you create or modify a mixer setup, all you have to do is **Save** it to be able to recall it at a later time.

Before you begin using PatchMix DSP, you need to set it up to be compatible with the other software applications you may be running. The most important consideration is your system sample rate. PatchMix DSP and any applications or other digital gear you are using must be set to the same sample rate. PatchMix DSP can run at 44.1k, 48k, 88.2k, 96k, 176.4k or 192k, but its complete set of features are only available at 44.1kHz or 48kHz. See Chapter 6 - Using High Sample Rates for complete details.

Once the sample rate is set, you can only easily switch between 44.1k and 48k. You cannot switch between 44/48k and the higher rates of 88k/96k/176k/192k. This is because the number of mixer inputs and outputs changes significantly at these high sample rates. In the case of such drastic sample rate changes, you must start a new session.

You can also set up an external sync source, thereby obtaining the sample rate from some other device or application. External sync can be obtained from the ADAT input, S/PDIF input or the Sync Daughter Card word clock. If the session is set at 44.1kHz or 48kHz and the external source is coming in at 96kHz (for example), the Sync Indicator will be extinguished (off), but PatchMix will attempt to receive the external data. The two units are NOT sample locked however, and you should correct this condition to avoid intermittent clicks in the audio. **Always check for the presence of the LOCKED indicator whenever you are using a digital interface.**

PatchMix DSP comes with several session templates to choose from so when you create a new session you can either create a "blank" session based around a designated sample rate, or select from a list of template starting points.

In a PatchMix DSP session the number of strips in the mixer is dynamically configurable. This allows you to create only those strips you need up to a maximum number determined by available DSP resources and available inputs.

▼ **Important:** *When using any form of digital input, you MUST synchronize the Digital Audio System to the external digital device (S/PDIF/ADAT) or synchronize all devices using Word Clock.*

New Session

You create a new session by clicking the "New Session" button in the PatchMix DSP main Toolbar. The following dialog box appears.

Select a Template or new Session at the desired sample rate

Session Description

Add your own comment or note about the Session

Check this if you want to edit the New Session.
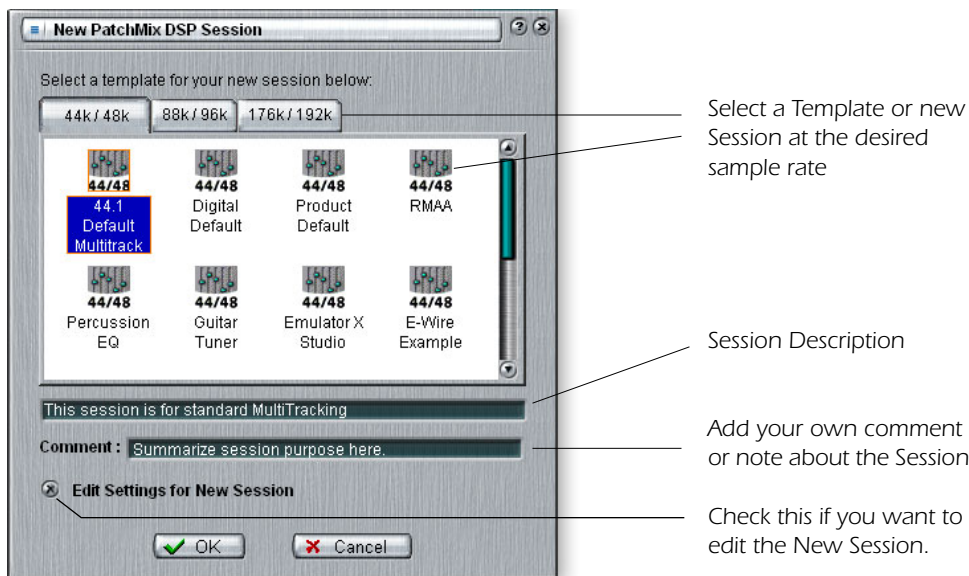
You can now select one of the factory template sessions. The factory templates are pre-programmed with specific setups such as audio recording or mixing. The selector tabs categorize Template Sessions into three groups based on sample rate, 44.1kHz/48kHz, 88.2kHz/96kHz, and 176.4kHz/192kHz.

You can create your own templates by simply copying or saving sessions into the "Session Templates" folder (Program Files\Creative Professional\E-MU PatchMix DSP\Session Templates). The system model number in parenthesis (1820) or (1212) must precede the template name in order to be recognized as a template.

The "Session Path" allows you to choose the destination for your Session. The default location is in the "My Sessions" folder within the "My Documents" folder.

There is also a Comment area that you can use to give yourself some clue as to what you were thinking when you created the session.

Open Session

To Open a saved session, click on the Open Session button. A dialog box appears allowing you to choose one of your saved Sessions to open. Choose one of your saved sessions and click on the Open button.

Save Session

To Save a session, click on the Save Session button. A Save dialog box appears allowing you to choose a location in which to save the current Session. The "My Sessions" folder is chosen by default.

Get in the habit of saving the session whenever you have created a special mixer setup. This will make you life much easier as you can recall a setup for many different audio modes such as: recording, mixing, special ASIO routings, etc.

❖ *Saving a session "defragments" the effect/DSP resources. If you have used all your effects and need another, try saving the session.*

Session Settings

System Settings
Pressing the Session Settings button on the toolbar brings up the System Settings window shown below. Click the tabs to select System, MIDI, or I/O options.
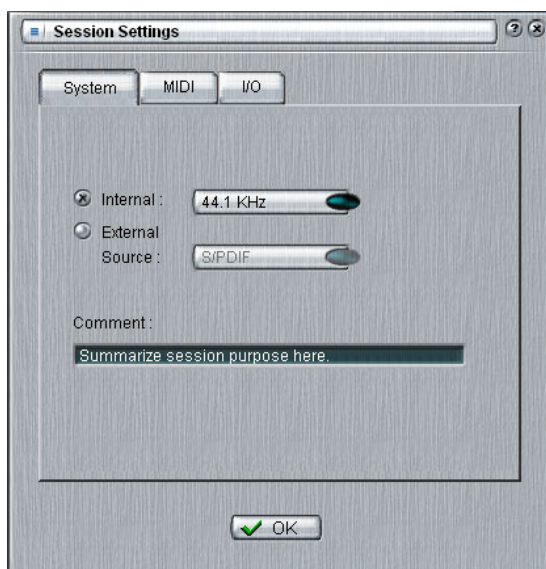
The System Settings include the following:

▪ **Internal/External Clock**	Selects between **internal** or **external** word clock source as the master clock source for the system
▪ **Sample Rate**	Selects the sample rate when using internal clock. Your choices are: **44.1kHz, 48kHz, 88.2 kHz, 96kHz, 176.4kHz, 192kHz.**
▪ **External Clock Source** (ext. clock only)	Select from: **ADAT, S/PDIF,** or **Word Clock** (Sync card only) as an external sample clock source.

▫ **Note:** if set to "External" without an external clock present, PatchMix DSP defaults to the internal 48kHz clock rate.

Using External Clock

Whenever you are using any digital I/O such as ADAT or S/PDIF, one of the digital devices MUST supply the master clock to the others. This master clock runs at the system sample rate (44.1k, 48k, 88.2k, 96k, 176.4k or 192 k) and can be distributed using a dedicated cable (word clock) or **embedded into a data stream such as S/PDIF or ADAT.** Common symptoms of unsynced digital audio include, random clicks or pops in the audio or failure of the digital stream to be recognized. **Always check for the presence of the "LOCKED" indicator whenever you are using a digital interface.**

▫ **Note:** The maximum supported sample rate for S/PDIF is 96kHz.

If an External Clock is interrupted or switched after the Session has been created (except between 44.1k <-> 48k), the "LOCKED" indicator will be extinguished and PatchMix will attempt to receive the external data. The two units are NOT sample locked however, and you should correct this condition to avoid intermittent clicks in the audio.

MIDI Settings

This option allows you to use either the MIDI In jack on the rear of the AudioDock or the Sync Card MIDI jack as a MTC Output. (The MTC Out only transmits MTC.)

▪ **Dock MIDI 2 In**	Selects the rear MIDI Input on the AudioDock as MIDI 2.
▪ **Sync Card Enabled**	Selects the Sync Card functions. This selection disables MIDI 2 In on the rear panel of the AudioDock. MIDI 2 Out on the rear panel of the AudioDock duplicates MIDI 1 Out in this mode.

▫ The SMPTE Input will not function and the Sync Card control panel will not be updated when Dock MIDI 2 In is selected.

Word Clock and SMPTE **Out** *will* operate with Dock MIDI 2 selected.

I/O Settings

You can set the level (-10dBV or +4 dBu) for each pair of analog outputs and the input gain setting for each pair of analog inputs.

An output setting of +4 provides the most output and is compatible with professional audio gear. Balanced output cables provide a +6dB hotter signal than unbalanced cables when used with balanced inputs. **Do NOT use balanced cables unless your other gear has balanced inputs.** See "Cables - balanced or unbalanced?" in the Appendix for more information.

Comparison of -10dBV & +4dBu Signal Levels

Consumer (unbalanced)   **Professional** (balanced)

+20 dBu   <– Clipping

Clipping –>   + 6 dBV = +8 dBu   **Headroom**

Headroom   + 2 dBV = +4 dBu

-10 dBV = -8 dBu

0 dBV = 1V RMS   0dBu = .777V RMS

An input setting of -10 is compatible with consumer audio gear and works best with low level signals. (-10dBV is approximately 12dB lower than +4dBu.) Choose the setting that allows you to send or receive a full scale signal without clipping.

Setting correct input and output levels is important! You can measure the level of an input by inserting a meter into the first effect location in the strip. Adjust your external equipment outputs for the optimum signal level. See "To Set the Input Levels of a Strip" for details.

❖ **Input too weak?**
Use -10 Input setting.

Output too weak?
Use +4 Output setting

Input Level Settings

Output Level Settings

E-MU 1010 Optical Input Select

E-MU 1010 Optical Output Select

S/PDIF Output Format Select
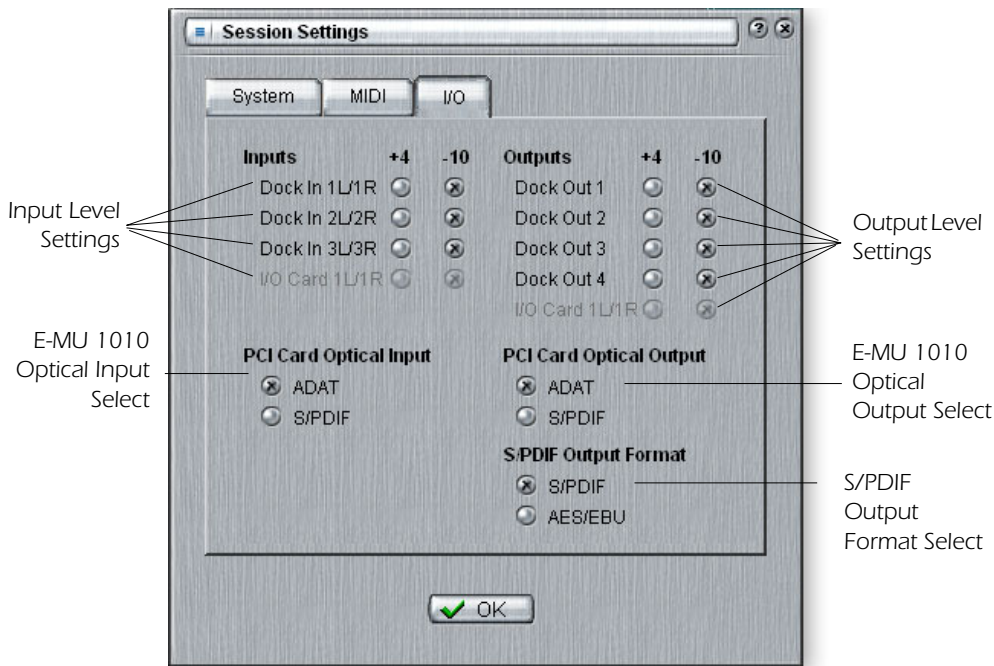
▪ **Inputs +4 or -10**	Selects between Consumer level (-10dBV) or Professional level (+4dBu) inputs. (Use the -10dBV setting if your input is too weak.)
▪ **Outputs +4 or -10**	Selects between Consumer level (-10dBV) or Professional level (+4dBu) outputs. (The +4 dBu setting outputs a hotter level.)
▪ **PCI Card Optical Input**	Selects between ADAT or optical S/PDIF for the 1010 PCI card ADAT Input. The coaxial S/PDIF input is disabled when S/PDIF optical is selected.
▪ **PCI Card Optical Output**	Selects between ADAT or optical S/PDIF for the 1010 PCI card ADAT Output. The coaxial S/PDIF Output is disabled when S/PDIF optical is selected.
▪ **S/PDIF Optical Format**	Selects between S/PDIF or AES/EBU format for S/PDIF. This sets the S/PDIF-AES status bit, but does not affect the signal level.

Input Mixer Strips

PatchMix DSP Input Mixer Strips are stereo except for the AudioDock Mic/Line inputs and the 0202 card inputs. Each input mixer strip can be divided into four basic sections.

▪ **Insert Section**	*Effects, EQ, External/Host Sends & Returns can be inserted into the signal path.*
▪ **Pan Controls**	*These controls position the signal in the stereo sound field.*
▪ **Aux Sends**	*Used to send the signal to sidechain effects or to create separate mixes.*
▪ **Volume Control**	*Controls the output level of the channel.*

Mono/Stereo

Input Type

Insert Section

Pan Controls

Aux Sends

Channel Volume Control

Mute/Solo Buttons

Scribble Strip
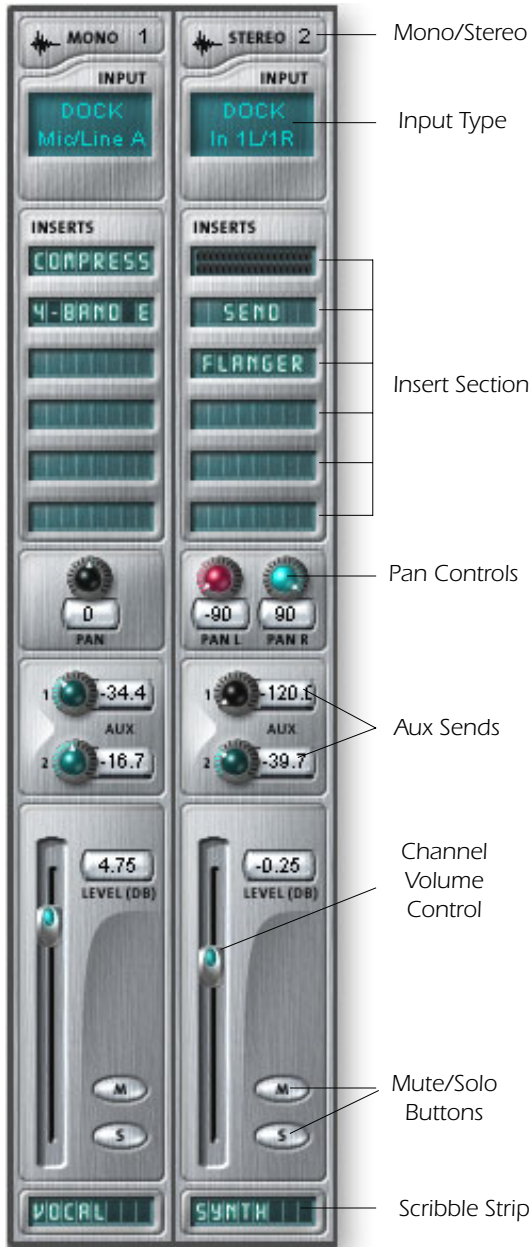
This screen shows a mono strip on the left and a stereo strip on the right.

Input Type
The very top of the strip is labeled mono or stereo and displays the type of the assigned input. Input mixer strips can be added as desired and can be configured to input the following:

- ▪ **Physical input**
  (Analog/SPDIF/ADAT).
- ▪ **Host Input**
  (Direct Sound, WAV, ASIO source)

Inserts
You can drag and drop effects from the Effects Palette or Right-click to insert a Physical or ASIO Send or Send/Return A Peak Meter, Trim Control or Test Signal can also be inserted by Right-clicking.

Pan Controls
These controls allow to you position the channel in the stereo sound field. Dual controls on stereo strips allow you to position each side independently.

Aux Sends
These controls send the signal to sidechain effect processors such as reverb and delay. They can also be used to create separate mixes for the artist or for recording.

Volume Control
Controls the output level of the strip into the main/monitor mix bus.

Mute/Solo Buttons
These convenient buttons allow you to solo or mute selected channels.

Scribble Strips
Click inside the scribble strip and type a name of up to eight characters.

❖ The **Input Type** will turn RED if the input is not available. (The AudioDock may be disconnected.)

❖ **Physical** input strips are shown with BLUE text.

❖ **Host** input strips are shown with WHITE text.

Mixer Strip Creation

PatchMix DSP is a dynamically configurable mixer. Each mixer session can contain an arbitrary number of strips up to a limit set by the number of available input sources and available DSP resources.

▶ **To Add a New Strip:**

1. Click on the "New Mixer Strip" button. <u>See Overview of the Mixer</u>

2. The New Mixer Strip Input Dialog appears:

❖ *Adding or deleting a strip "defragments" the effect/DSP resources. If you have used all your effects and need another, try deleting an unused strip.*
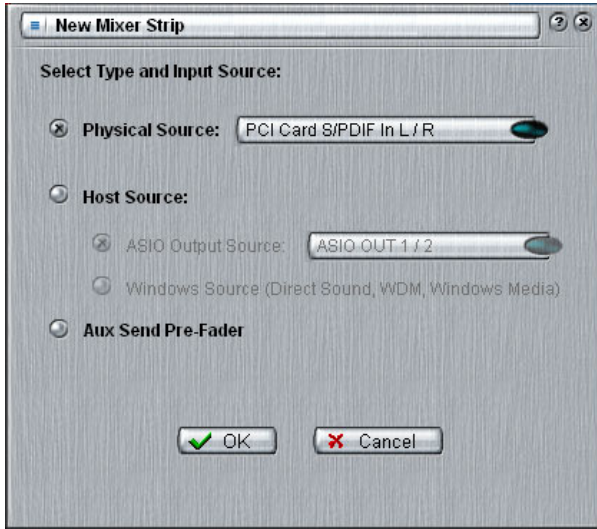
3. Select the desired input to the mixer strip from the following choices:

▪ **Physical Source:**	Analog or digital input (Analog, ADAT, S/PDIF)
▪ **Host - ASIO Source input**	Streaming audio from an ASIO software application.
▪ **Host - WAVE input**	Window sound sources — WAVE, WDM, CD

❖ **CDs & MP3s:** *The* **WAVE 1/2** *strip is used to playback CDs, Windows Media Player, and Direct Sound.*

Mixer Strip Type	Function
Physical: I/O Card In	24-bit monophonic analog input from the 0202 Daughter Card.
Physical: Dock Mic/Line	24-bit monophonic analog input from the AudioDock.
Physical: Dock In	24-bit stereo analog input from the AudioDock.
Physical: PCI Card S/PDIF	2 channel digital audio from the S/PDIF input on the E-MU 1010 card.
Physical: PCI Card ADAT	2 channel digital audio from the ADAT input on the E-MU 1010 card
HOST SOURCE	**Function**
Host ASIO Output Source From software application	2 channel digital audio from an ASIO source (software app). ASIO: 1/2, 3/4, 5/6, 7/8, 9/10, 11/12, 13/14 … 31/32
Host Windows Source From Windows	Direct Sound, WDM, Windows Media *(Sound generated or handled by Windows.)* WAVE 1/2 - Default stereo source such as game sound, CD player, beep sounds, etc. WAVE 3/4, WAVE 5/6, WAVE 7/8 - Additional WDM channels

4. Select **Pre-Fader** Aux Sends or leave the box unchecked for **Post-Fader** Aux Sends.

5. Click **OK** to create a new strip or **Cancel** to cancel the operation.

▶ **To Delete a Mixer Strip:**

1. Click the top of the mixer strip you wish to delete. A red border appears around the strip, indicating that it is selected.

2. Click on the Delete Mixer Strip button, or right-click and choose **Delete**, or use the **Delete key** on the PC keyboard. See Overview of the Mixer

Multichannel WAVE Files

2 channels of WAVE recording and **8 channels of multichannel WAVE playback are supported**. The WAVE channels are available for the following types of WDM devices:

- Classic MME

- DirectSound

- Direct WDM / Kernel Streaming (KS)

DirectSound and the **WDM/KS** interfaces allow up to Eight channels of Wave Out while the classic MME interface only exposes 2 channels.

The WAVE channels operate at all sample rates. For additional information about WDM behavior at high sample rates, see _"WDM Recording and Playback Behavior"_.

192kHz/96kHz DVD-Audio disks are protected against digital copying. However, most DVD-Audio disks contain duplicate 48kHz audio tracks which will play back on the Digital Audio System.

Windows Media Player/DVD/Surround Sound Playback

Select **DirectSound** as the output format when using Windows Media Player and other DVD player applications.

Eight channel WAVE playback supports 5.1, 6.1 or 7.1 surround audio.

The chart below shows how to connect the outputs for surround sound playback. Use outputs 1-3 for 5.1 surround; use outputs 1-4 for 7.1 surround.

Multichannel WAVE to Surround Sound Speaker Channels
(using the 5.1 DVD Playback Session)

WAVE Strip	Surround Channels	1/4" Outputs	1/8" Outputs
E-DSP WAVE 1/2	Front Left / Front Right	1L = FL 1R = FR	1 (Tip = FL Ring = FR)
E-DSP WAVE 3/4	Center / Subwoofer	3L = C 3R = Sub	3 (Tip = C Ring = Sub)
E-DSP WAVE 5/6	Rear Left / Rear Right	2L = RL 2R = RR	2 (Tip = RL Ring = RR)
E-DSP WAVE 7/8	Side Left / Side Right	4L =SL 4R = SR	4 (Tip = SL Ring = SR)

Insert Section

The **Insert Section** is next in line. PatchMix DSP effects can be selected from the Effects Palette and dropped into the insert locations. See "The Effects Palette". Any number of effects can be inserted in series.

The Inserts also have the unique ability to patch into ASIO/WAVE and external equipment. ASIO/WAVE Sends, External Sends and External Send/Returns can be dropped into the insert section to route the signal anywhere you want.

The Insert/Patch Bay is incredibly flexible. Want to send the input of the strip to your audio recorder? Simply insert an ASIO send into the insert section and select the ASIO pair you want. That's it! That input is now available in your ASIO software.

❖ You have to create an ASIO strip or ASIO Send in order to activate these ASIO channels in your software.

The following types of inserts can be selected.

|	
Hardware Effect	Reverb, EQ, Compressor, Flanger, etc. using PatchMix DSP's effects which do not load your CPU.
Host ASIO Send	Splits off the signal and sends it to an ASIO host input such as a software audio recorder or anything that uses ASIO.
ASIO Direct Monitor	Sends the signal to a selected ASIO host input, then returns a selected ASIO host output to the chain. Use for recording with "Direct Monitoring"
Ext. Send/Return	Sends signal to a selected external output, then returns it to the chain via a physical input.
External Send	Sends the signal to an external output. See "To Add a Send Insert:".
Peak Meter	Peak meters allow you to monitor the signal level anywhere in the chain. See "Meter Inserts".
Trim Pot	You can insert a gain control with up to 30 dB of gain or attenuation. A peak level meter and phase inverter are also included. See "Trim Pot Insert".
Test Tone	This special insert outputs a calibrated sine wave or noise source, which can be used to track down audio problems. See "Test Tone/Signal Generator Insert".

Working with Inserts

The Inserts are one of most powerful features of the PatchMix DSP system as they allow you to configure the mixer for a wide variety of applications.

▶ **To Add an Effect to an Insert Location:**

1. Press the **FX** button. The effects palette appears.

2. The effects are organized into categories. Click on a folder to open it.

3. Select the effect you want, **drag it** over the insert section, then **drop it** into an insert location.

4. To rearrange the order of effects, simply **drag and drop** them into the desired order.
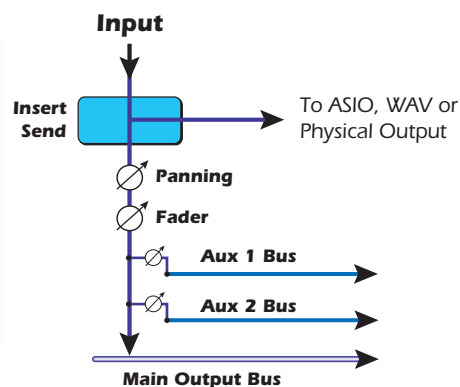
The Insert Menu

Right-Clicking over the insert section brings up a pop-up selection box containing various insert options to help you control and manage your inserts.
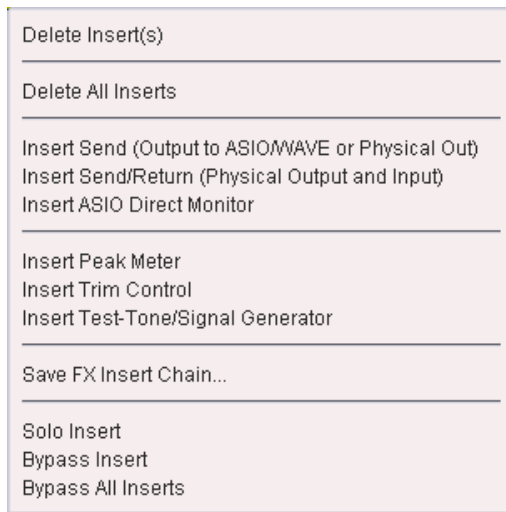
▶ **To Add a Send Insert:**

This type of insert send *splits* the signal at the insert point and sends it out to the selected destination. (An "ASIO Send" becomes an input on your recording application, a "Physical Out" goes to a pair of output jacks. the signal also continues down the strip to the Aux Sends and main mixer outputs.)

1. Right-Click over the Insert section. A pop-up dialog box appears.

2. Select "Insert Send" (to ASIO/WAVE or physical output) from the list of options. The following dialog box appears.
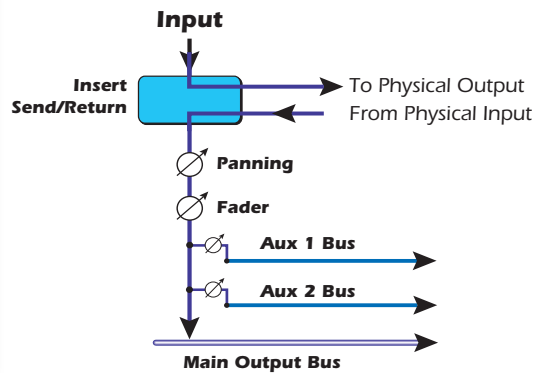

3. Choose one of the Send Outputs. Click on a destination to select it.

4. Click O**K** to select the output or **Cancel** to cancel the operation.

▶ **To Add a Send/Return Insert:**

This type of insert send *breaks* the signal at the insert point and sends it out to the selected destination such as an external effect processor. A return source signal is also selected which *returns* the signal to the channel strip after processing.

1. Right-Click over the Insert section. A pop-up dialog box appears.

2. Select "Insert Send/Return (Physical Output and Input)" from the list of options. The following dialog box appears.
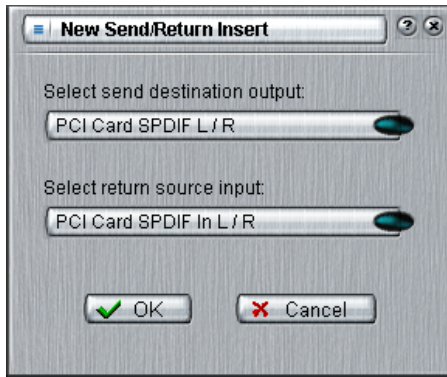
▼ If the source or destination you want to use is not available in the list, they are probably already being used elsewhere. Check the input Strips, Inserts and Output Assignments.

3. Choose one of the Send Outputs. Click on a destination to select it.

4. Choose one of the Return Inputs. Click on a source to select it.

5. Click **OK** to select the Send and Return, or **Cancel** to cancel the operation.

Using External Sends & Returns

An External Send/Return *breaks* the signal at the insert point and sends it out to the selected external destination such as a compressor, a stereo effect processor or any other audio device. Any physical input or output can be used.
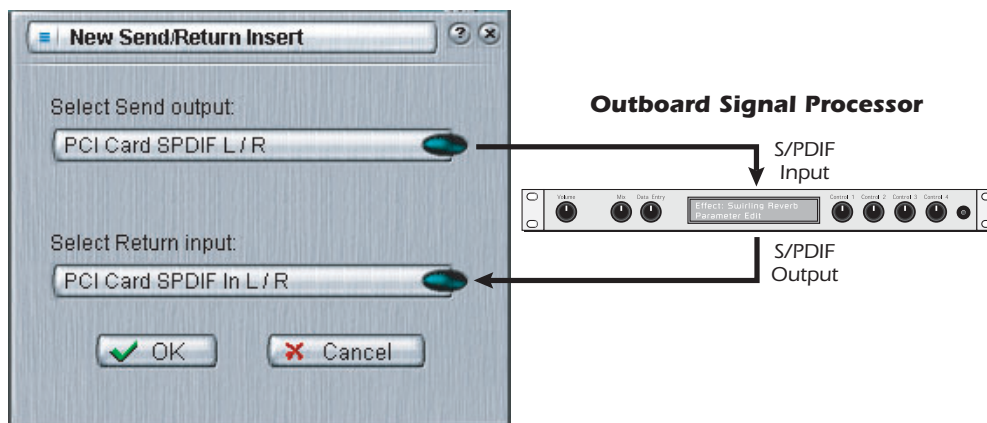
The Send/Return Inserts allow you to connect your favorite external analog or digital gear.

▶ **To Insert an External Signal Processing Device:**

1. Right-click on the insert chain and select the **Insert Send/Return** (Physical Output and Input). The popup dialog box shown above appears.

2. Select the output and input you wish to use.

3. Connect your external audio gear to the appropriate analog or digital connectors.

4. If you're using the analog outputs and inputs, you may have to adjust the levels in the I/O Session Settings. See page 31. You can insert a level meter after the Send/ Return to make sure you're getting a strong signal back from the external device.

5. If you're using a digital Send/Return, make sure that the external digital device is synchronized to the digital input. Otherwise, poor audio quality will result. *(Alternatively, you can synchronize PatchMix DSP to your external device in the System Session Settings. See page 30.)*

ASIO Direct Monitor Send/Return

This type of insert send *breaks* the signal at the insert point and sends it out to the selected ASIO Host Input destination (such as Cubase or Sonar). A return source signal is also selected which *returns* the signal to the channel strip from an ASIO Host Output.

The ASIO Direct Monitor Send/Return is unique in that it utilizes ASIO 2.0 zero-latency monitoring. **In order to utilize this feature, Direct Monitoring must be enabled in the audio recording application.**

While recording, the Direct Monitor Send/Return routes the signal to the recording application, but monitors directly from the input to eliminate latency. During playback, the recording application automatically switches the Direct Monitor Send/Return to monitor the recorded track.

Recording   **Playback**
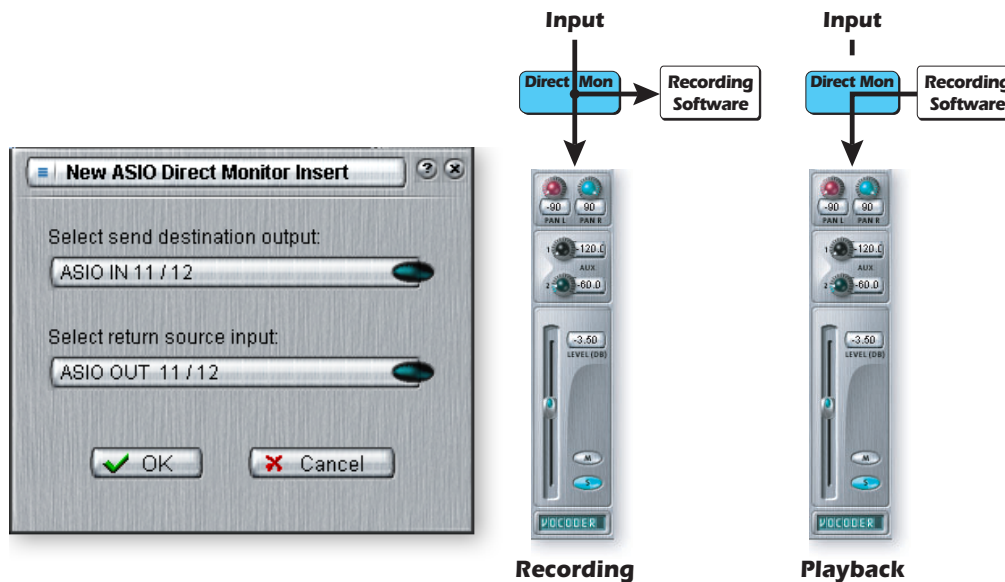
The Direct Monitor Send/Return also allows the recording application to control volume and pan. Normally when using direct monitor recording you'll want to control the volume and pan from the recording application. In this case, set the PatchMix DSP stereo pan controls hard left and right, mono pan controls to center, and the fader to 0dB.

▶ **To Add an ASIO Direct Monitor Send/Return:**

1. Right-Click over the Insert section. A pop-up dialog box appears.

2. Select "Insert ASIO Direct Monitor" from the list of options. The following dialog box appears.

3. Choose one of the Send Outputs. Click on a destination to select it.

4. Choose one of the Return Inputs. Click on a source to select it.

5. Click **OK** to select the Send and Return, or **Cancel** to cancel the operation.

Meter Inserts

Keeping track of signal levels is important in any audio system, be it analog or digital. You want to keep the signal levels running as close to maximum in order to achieve high resolution and low noise. On the other hand, you don't want the signal level so high as to cause clipping. To help you maintain optimum signal levels, we have included Peak Level Meters, which can be dropped into any insert location.

The insert meters are of the "peak hold" type. The topmost bar in the meter holds its highest level for a second to let you see transients that would otherwise be too quick for the eye. A numeric readout above the meter shows the peak-hold level in dB.

The peak meters are also color-coded to indicate the signal strength. The chart below outlines the meanings of the colors. Avoid lighting the topmost red bar, as this indicates distortion of the signal. Click on the clip indicator to turn it off.

Meter Color	Indicates
▪ Red	*Indicates signal clipping.*
▪ Yellow	*Good strong signal level.*
▪ Green	*Signal is present.*

One of the most obvious uses of the insert meters is to set input levels. On the analog inputs, the analog-to-digital converter (ADC) is one of the most critical points in the signal path. You want the input signal level to drive the 24-bit ADCs into their optimum range without clipping. A reading of 0dB on an input meter indicates signal clipping.
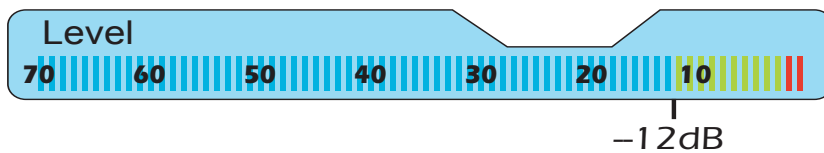
Each bar of the meter equals 1dB. The yellow bars begin at -12dB below full scale.

The insert meters are also useful to monitor incoming digital signals such as ADAT, ASIO or S/PDIF to make sure the mixer is receiving a proper signal level. They're also great for troubleshooting, since you can place them virtually anywhere in the mixer.

▶ To Insert a Meter

1. Right-Click on an Insert location of the mixer strip. A pop-up dialog box appears.

2. Select **Insert Peak Meter**. A stereo peak meter appears in the insert location.

3. Select **Effect** in the Main Section. The meters are now shown in high resolution in the TV screen.

To Set the Input Levels of a Strip

1. Select the topmost Insert location on a mixer strip and insert a meter (see above).

2. Left-click on the meter insert to see the meter in the TV screen.

3. Feed your audio signal to the input of the mixer strip. The meter should now show the signal level.

4. Adjust the output level of the external device (synthesizer, instrument, preamp, etc.) feeding the AudioDock or 0202 Daughter Card. The meter should be in the yellow region most of the time with occasional forays into the red. If the clip indicator ever comes on, reduce the signal level.

❖ **Input too weak?**
Use -10 Input setting.

Output too weak?
Use +4 Output setting

5. Each analog input pair has its own Input Pad (-10dBV or +4dBu) which controls the input signal range. Changing the I/O settings can add or subtract 12dB. Check these settings if you cannot set the proper input level. See I/O Settings.

Making the Best Possible Recording

Making a good digital recording is easier than ever thanks to the high resolution 24-bit A-D converters on your Digital Audio System. These converters are much more forgiving than the 12-bit or 16-bit converters of the past. Even so, to get the best performance possible, you'll need to follow a few basic guidelines.

First, whenever you input an analog signal to the Digital Audio System, make sure that you're feeding the A-D converters with an optimum signal level. The quality of a digital recording is directly related to the signal level you feed into the A-D converters. If the analog input level is set too low, you lose resolution—if it's set too high, the A-D converters will clip.

To measure the input level, simply add an insert meter to the channel strip in PatchMix DSP. These meters are accurately calibrated to display 1dB for each bar on the meter. You can enlarge the meter view by clicking on the insert meter in a strip and selecting the "Effect" button at the top of the TV screen.

The "I/O Settings" in the Digital Audio System allow you to set the input levels to -10dBV (consumer equipment level) or +4dBu (professional equipment level) for each analog input. This control sets the overall input level to match your other gear, but to get the best possible recording you need to fine tune the level further.

In order to supply the correct input level, you'll need to adjust the output of your analog source (electric instrument or preamp) so that the input level comes close to 0dB without ever going over.

Play your input source signal while watching the insert meter in the strip. The signal should go into the yellow area frequently, but never into the red. Adjust the level of your source until you have a good level. If the signal is way too strong or too weak, you may have to go back and adjust the I/O Settings. **Choose "-10" if the input signal is too weak and "+4" if the signal is too strong.**

Digital audio has NO headroom past 0dBFS (FS = Full Scale) and will "hard clip" if the signal exceeds 0dB. Hard clipping sounds bad and will ruin your recording. Hard clipping occurs because at 0dBFS, all 24 bits are turned on and the A-D cannot measure any higher level. Analog tape, unlike digital, can be driven past 0dB, although with some degradation of the signal.

The Digital Audio System includes Insert "Trim Pot" controls, but they adjust the signal level after the signal has been digitized and will not recover any lost resolution. It's far better to set the input level correctly in the first place. Trim Pots can be used in emergency situations if there's no other way to get a hot signal in, but they were designed to adjust levels feeding effect plug-ins.

Trim Pot Insert

The Trim Pot Insert allows you to adjust the level of a signal in an insert location. The trim pot provides up to ±30dB of gain or attenuation and a phase inverter. The trim pot also has a built-in stereo peak meter after the control.
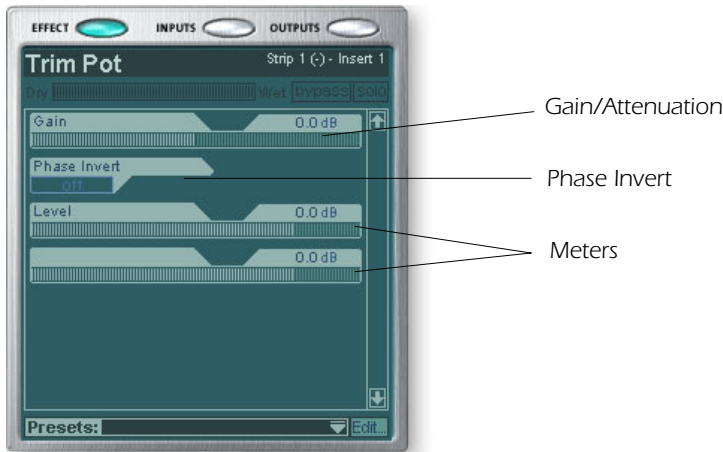
You might use a trim pot to boost or attenuate a send or return from an external effect, or to drive an effect device. Certain effects such as the Compressor, Distortion, and Auto-Wah are very level dependent and like to see a good, strong input signal. If you are working with a weak signal, you can improve the performance of these effects inserting a trim pot and boosting the gain.

Trim pots can be used to boost the level of analog line level inputs, but it's much better to boost the signal level *before* the A/D converters in order to get maximum resolution and signal-to-noise ratio from the converters.

The phase invert switch inverts the polarity of the signal. It is generally used to correct for balanced lines and mics that are wired backwards.

Test Tone/Signal Generator Insert

The test tone/signal generator insert is a handy troubleshooting aid which outputs a calibrated sine wave, white noise or pink noise. This tool, in combination with an insert meter, allows you to accurately measure the signal gain or attenuation of an internal or external device. The test tone can also be quite handy for tuning up musical instruments.

The sine wave oscillator frequency is variable from 20Hz-20kHz. The level is variable from off to +30dB.

White noise is a mixture of all frequencies in the audio spectrum at the same average level (analogous to white light in the visible spectrum).

Pink noise provides equal power distribution per octave. (White noise has more power in the higher octaves.) Pink noise and white noise are useful as wideband sound sources.

❖ Musical Note Freq.
A = 440 Hz
B = 493.88 Hz
C = 523.25 Hz
D = 587.33 Hz
E = 659.26 Hz
F = 698.46 Hz
G = 783.99 Hz

Managing Your Inserts

▶ **To Delete an Insert:**

1. Right-Click over the Insert you wish to delete. A yellow line around the insert location indicates that it is selected. A pop-up dialog box appears.

2. Select **Delete Insert** to remove the selected insert or select **Delete All Inserts** to remove all inserts.

3. The insert(s) are deleted from the insert chain.

❖ **Tip:** Select the Insert and press the **Delete** key to delete the plug-in from the strip.

▶ **To Bypass an Insert:**

Inserts can be bypassed if you want to temporarily hear the audio without the effect or insert. Bypass can also be used to turn off a Send Insert.

Method #1

1. Click on the Effect (in the Insert section) and select **Effect** in the TV display.

2. Click the **Bypass** button.

Method #2

1. Right-Click over the Effect you want to bypass (in the Insert section). A pop-up dialog box appears.

2. Select **Bypass Insert** from the list of options.

▶ **To Bypass All Inserts:**

All Inserts in a strip can be also be bypassed with a single command.

1. Right-Click over the Effect you want to bypass (in the Insert section). A pop-up dialog box appears.

2. Select **Bypass All Inserts** from the list of options.

▶ **To Solo an Insert:**

Inserts can also be soloed. Solo bypasses all the other inserts in the strip and allows you to hear only the soloed effect. This feature is very useful when adjusting the effect parameters.

Method #1

1. Click on the Effect (in the Insert section) and select **Effect** in the TV display.

2. Click the **Solo** button.

Method #2

1. Right-Click over the Effect you want to Solo (in the Insert section). A pop-up dialog box appears.

2. Select **Solo Insert** from the list of options.

Aux Section

The Auxiliary Sends tap the signal from the channel strips and sum them together before sending the mix to the Auxiliary Effects section. In a traditional mixing console, aux sends are used to send part of the signal to outboard effect devices, then return the effected signal back into the mix using the effect returns. This is called a sidechain routing because the aux signal takes a detour through the effects before being summed back into the main mix. Sidechain effects are usually effects that you might want applied to several channels, such as reverb.

Incidentally, the wet/dry mix of effects in the Aux Sends should normally be set to 100% wet. This is because you will be adjusting the effect amount using the Aux Return control. If you have more than one effect in an Aux Bus, ignore the preceding advice as the wet/dry controls can be used to mix the amounts of your multiple effects.

The Aux 1 & 2 buses can also be used as additional submix output buses just like the main output. Simply drop an ASIO or External Send Insert into the chain and the stereo bus is sent. Turn down the Return Amount if you don't want the submix to be combined into the main mix.

Aux Send and Return values can also be changed by typing directly into the displays.
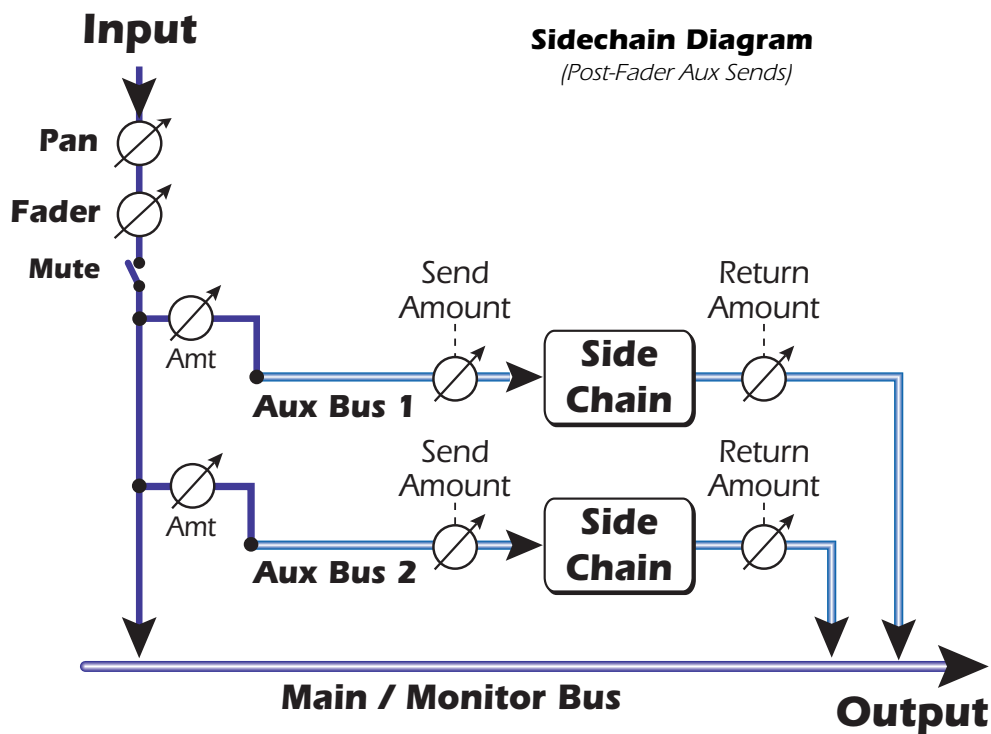

Other Uses of the Aux Sends

You can think of the Aux Sends as two extra mixing buses because that's exactly what they are. These two mixes can be routed anywhere, such as to a physical output or an ASIO pair. You could route one of the Aux buses to the Monitor out to create a monitor mix while sending the main mix off to your audio recording software.

Pre or Post Fader Aux Sends

When you create a New Mixer Strip you have the option to place both Aux Sends *after* the channel volume fader and mute control or you can place them *before* the fader and mute. Post-Fader turns down the send level as you lower the volume of the strip. With Pre-Fader selected, you may still hear the effected signal returning from one of the Aux Buses with the volume fader turned down.

With the Pre-Fader box selected, the Aux Send levels are completely unaffected by the Level Fader and Mute settings. The Pre-Fader setting allows you to create two completely different mixes using the Aux Buses since the signal levels of this mix won't be affected by the fader settings.
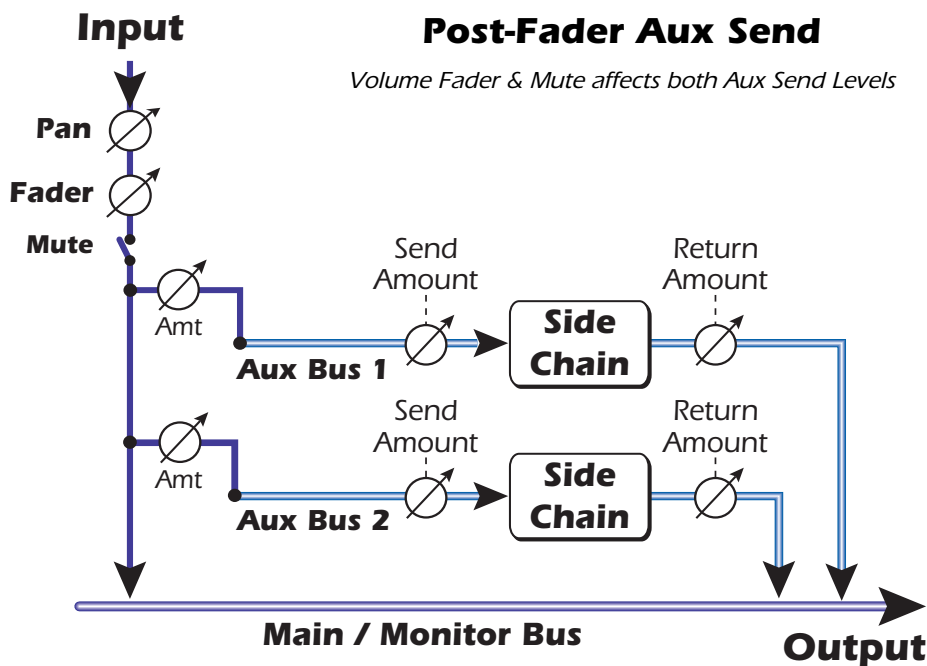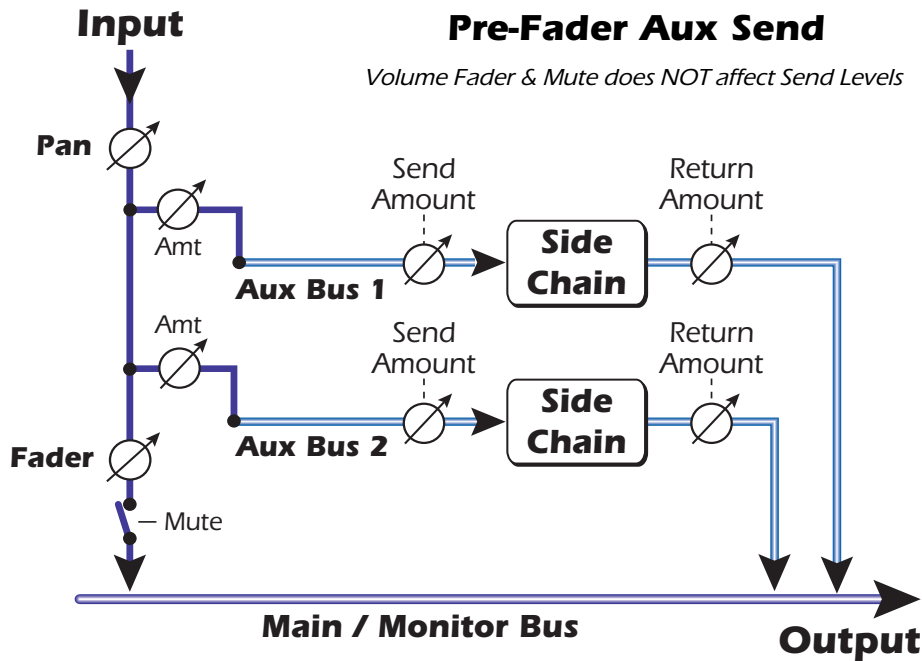
▼ In order to change a strip from pre-fader to post-fader or vice-versa, you have to delete the strip and create a new one.

Pre-Fader Aux Send
Volume Fader & Mute does NOT affect Send Levels

Post-Fader Aux Send
Volume Fader & Mute affects both Aux Send Levels

Level, Pan, Solo & Mute Controls

Pan Controls

Aux Send Amount Controls

Level Control

Mute & Solo Buttons

Scribble Strip

The **Pan** control comes before the Level Control and Aux Sends in the signal flow. On stereo strips we use an unconventional pan section with two pan pots – one for the left part of the signal and one for the right part of the signal. This feature allows you to independently position both sides of the stereo signal. A conventional stereo balance control only allows you to turn down one side or the other.

The **Mute** button does just what you would expect—press the button and the sound from that channel is cut off. Pressing the Solo button while the Mute button is pressed allows you to hear the channel until solo is turned off.

The **Solo** button allows you to listen to only that channel while muting the rest of the mixer's output. If multiple solo buttons are pressed, you will hear all soloed channels and the non-soloed channels will all be muted.

The mute status is remembered if a muted channel is soloed. When the channel solo is turned off, the channel reverts to being muted.

The **Level Control** for the strip is an attenuation control that can also provide up to +12dB of gain. 0db is the unity gain setting. You can also type numeric values into the displays to set the level.

At the very bottom is the **Scribble Strip** text area, into which you can type any short piece of text, thus naming the strip, i.e. "vocals", "bass", "drums" and so on.

Main Section
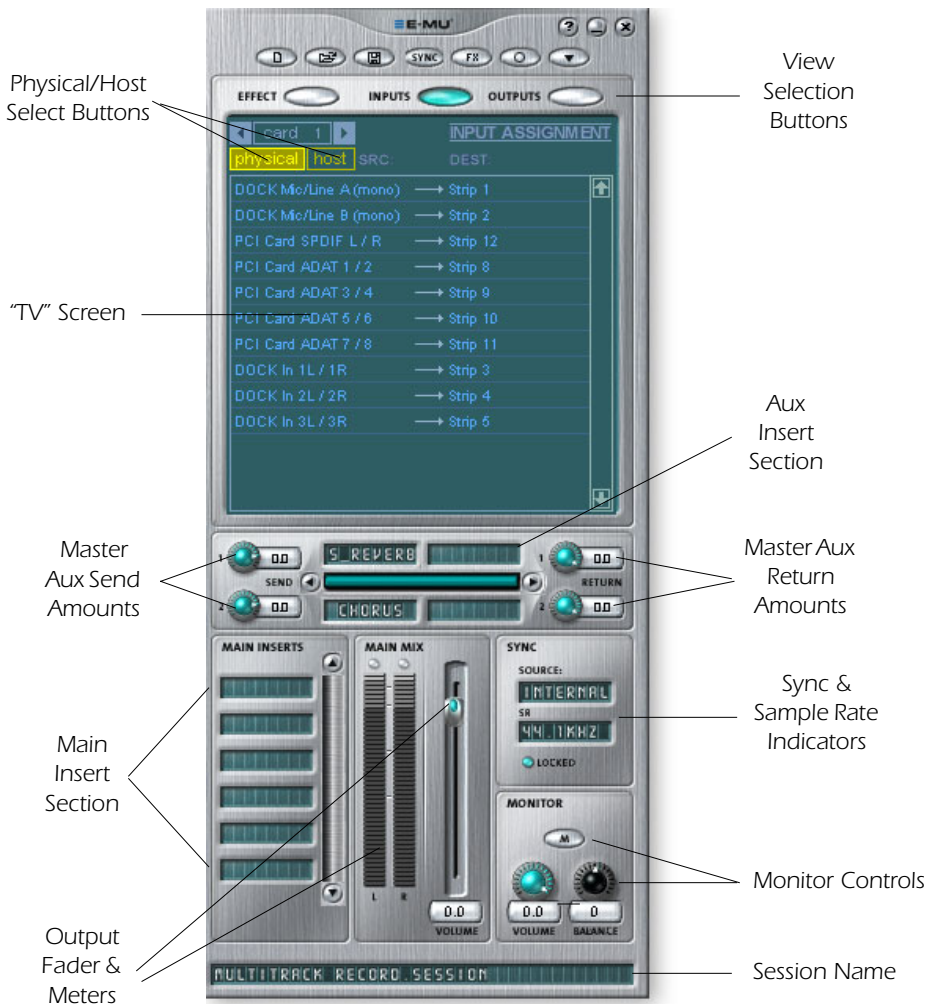
The main section contains all controls for controlling the main mix elements as well as a "TV screen" for viewing the parameters of the current selected insert.

The three buttons across the top of the main section select what is shown on the TV display. Input and output routings are graphically displayed. When an insert is selected (by clicking on the insert), the screen shows the available parameters for the currently selected insert.

Below the TV screen is the Aux Bus section where effects, effects chains or other inserts can be assigned to the two aux buses. Send and return levels can be individually controlled for each of the two Aux Buses.

The Aux 1 and Aux 2 buses are fed by the two Aux Sends on each mixer strip. The Master Send Level control on Aux bus 1 and 2 can be used to attenuate or boost the signal going into the Auxiliary Inserts. There is also a Master Return Level to control the amount of the effected signal that will be returned into the main mix.

The Main Bus can also have a chain of effects inserted. (You might put an EQ here to equalize your entire mix or add an ASIO or WAVE send to record the mix.) Note that the Main Output level control comes before the Monitor Level so that you can control the monitor level without affecting the level of your recording mix or main mix. There is a stereo peak meter that indicates the signal strength for the main mix.

The Monitor section has a volume, balance, and a mute control to cut off the monitor output.
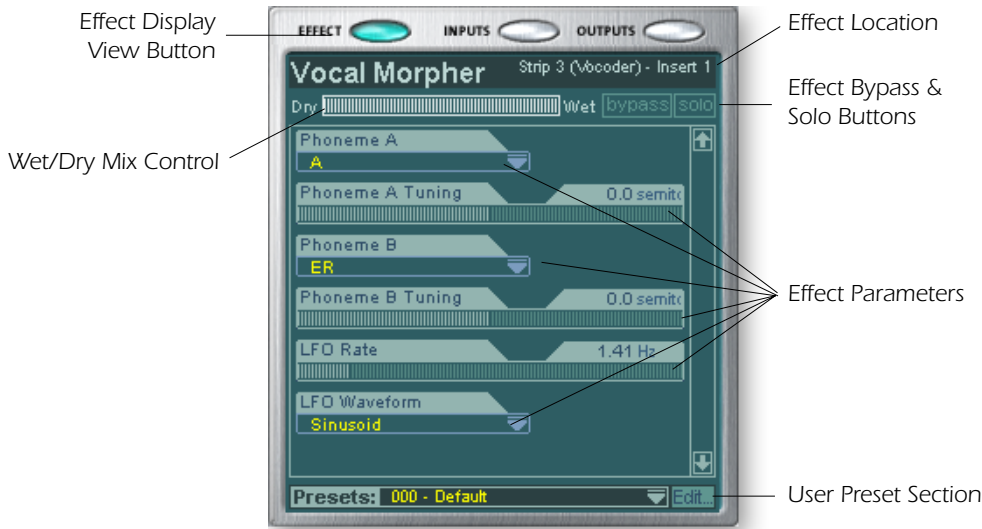
TV Screen & Selectors

The "TV screen" at the top of the main section is a multi-function display and control center for the input and output routings and effect controls. The three buttons at the top of the display select the current function of the display—Effect, Inputs or Outputs.
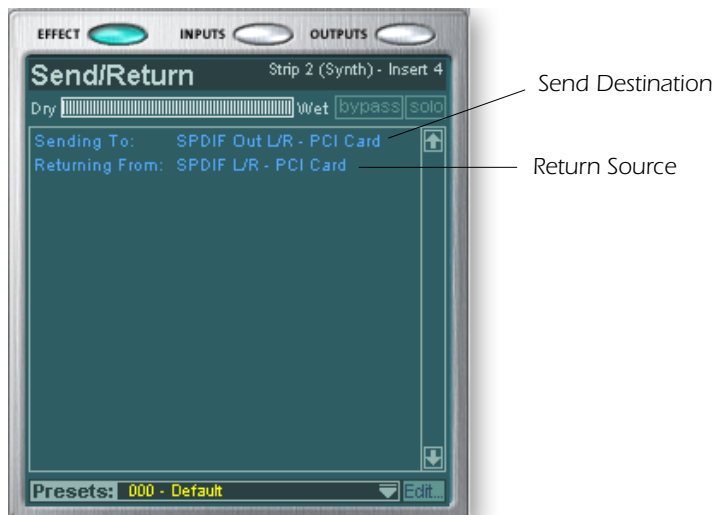
Effect

Select the **Effect** display view in the main section, then click on an Effect Insert to display the effect parameters. If an insert effect is not selected, the display will read "No Insert".

Most effects have a wet/dry mix parameter to control the ratio of effect to plain signal. The wet/dry setting is stored with the effect preset. The parameter set varies with the type of effect. See "List of Core Effects" for detailed information about the individual effects.

:: **Note:** *Effects have to be placed into an insert location before you can program them.*

When a Send or a Send/Return insert is selected with the effects display enabled, the TV screen shows you where the Send is going and where the Return is coming from. The bypass or solo buttons at the top of the display are available for Send/Return type inserts only.

Input

Selecting the **Input** display view shows a graphic representation of the PatchMix DSP Mixer inputs. This screen is only a display, unlike the Effects and Outputs screens, which allow you to make routing changes. Input routing changes are made by adding mixer strips. See Mixer Strip Creation.

The input routings are divided into two categories: Physical Inputs and Host Inputs. Select either category by clicking on the Physical or Host button.

❖ *The Input and Output displays make it much easier to understand the signal routings of a complex mixer setup.*

Physical Input Display   Host Input Display

❖ **Tip:** *Clicking on any of the input routings in the TV display highlights the corresponding mixer strip.*
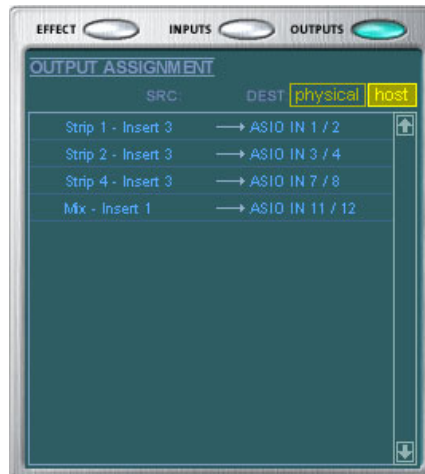
Output

Selecting the **Output** display view shows a graphic representation of the PatchMix DSP Mixer outputs. The output routings are divided into two categories: Physical Outputs and Host Outputs. Select either category by clicking on the Physical or Host button.

Physical Output Display   Host Output Display

The Host Output display shows all the Insert Routings in addition to the Main Mix and Monitor out routings. Click on the desired row to make or break a physical output connection.

The **Physical Output** screen displays and allows you to connect the Main and Monitor outputs of the mixer to "physical" analog or digital outputs. Click on the box in the mix or monitor area to make (or break) a connection.

The **Host Output** screen displays and allows you to view the Host (ASIO or WAVE) outputs of the mixer. See "Insert Section" for information on how to connect the inserts.

Auxiliary Effects & Returns

The section immediately below the TV Screen is where you assign the Auxiliary Effects. In a traditional mixing console, auxiliary effects sends are used to send part of the signal to outboard effect devices, then return the effected signal back into the mix using the effect returns. This is called a sidechain routing because the aux signal takes a detour through the effects before being summed back into the main mix.

Sidechain effects are usually effects that you might want applied to several channels, such as reverb. Effects such as EQ and compressors are usually NOT used as sidechain effects because they can cause unpredictable results when returned to the main bus.

❖ *The Wet/Dry mix setting in the effect should normally be set to 100% when the effect is inserted as a sidechain effect. This is because the Aux Return Amount will control the wet/dry mix.*

Sidechain Diagram
(Post-Fader Aux Sends)
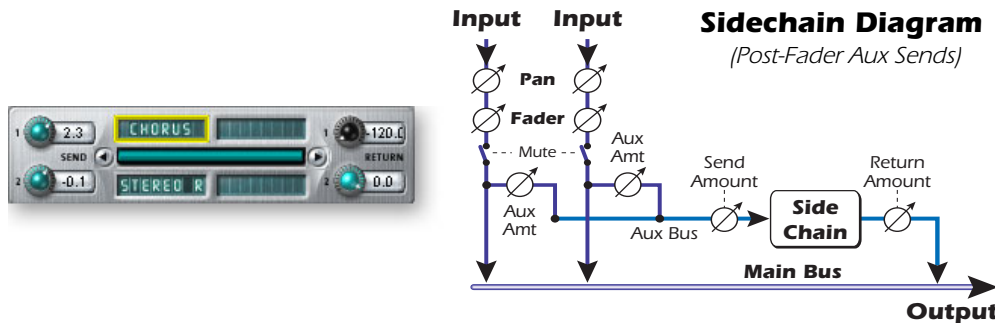
You can also use the Auxiliary Sends as two extra mix buses. By turning the Aux Return amount all the way down and dropping an Insert Send into the chain, you can send the Auxiliary bus to any output you wish. See "Insert Section" for more information.,
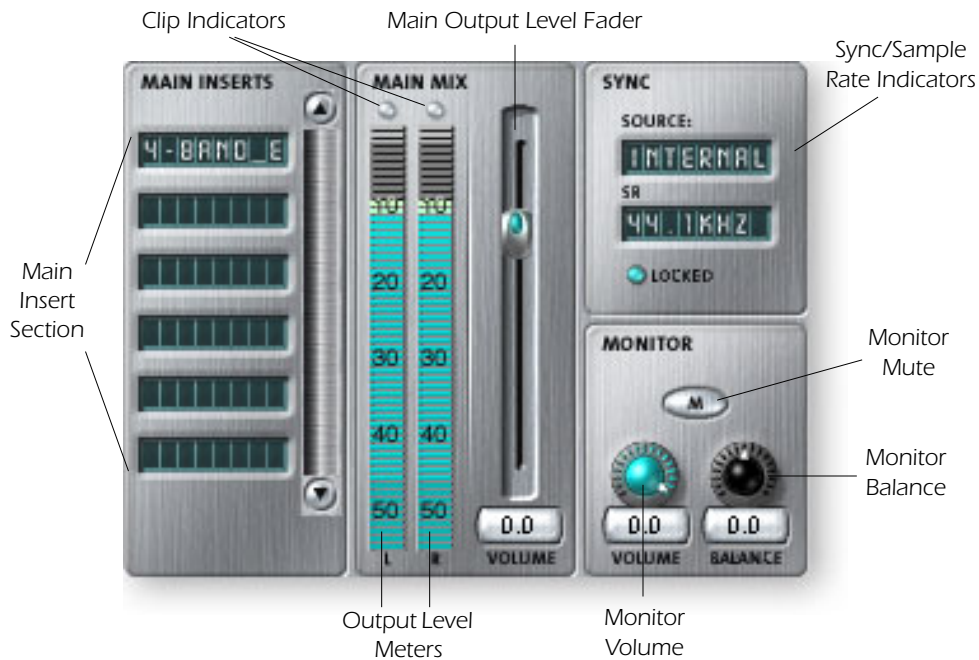
Sync/Sample Rate Indicators

The Sync/Sample rate Indicators show the current session's sample rate and whether it is internal or slaving to an external source. The LEDs indicate which sample rate is currently in effect. If an external source is being used, the Source display reads "EXTERNAL".

When slaving to an external master source, the clock may drift slightly or change dramatically (i.e. abrupt sample rate change or unplugging of physical master source). PatchMix DSP is tolerant to minor drifting within the supported rates of 44.1k, 48k, 88k, 96k, 176k and 192k, but if the sample rate drifts out of this range the "LOCKED" LED will extinguish.

If the external clock source makes a radical sample rate change from the lower rates of 44.1k/48k to a higher rate (or between any of the higher rates), the hardware automatically switches to internal 48kHz clock until the proper external clock is restored. The "LOCKED" LED will be off and the two units are NOT synchronized. Always check the "LOCKED" LED when using an external clock source.

Output Section

Clip Indicators · Main Output Level Fader · Sync/Sample Rate Indicators · Main Insert Section · Output Level Meters · Monitor Volume · Monitor Mute · Monitor Balance

Main Inserts

The main inserts allow you to apply effects to the main stereo signal coming out of the mixer (both mains and monitor). You might want to apply EQ or a compressor here. These inserts work just like the other insert locations—just drag and drop effects from the palette or right-click and add Sends, Sends/Returns. etc. Refer to the Mixer Block Diagram

Main Output Fader

The main output fader controls the level of the main output (and the Monitor output as well since it is downstream from this control). The normal setting for this control is at unity or 0dB, but the control allows you to add up to +12dB of gain. High output levels may cause clipping on outboard amplifiers or other equipment.

Output Level Meters

This stereo bar-graph meter reflects the digital level at the output of the mixer. The topmost red bar represents 0 dB or a full-scale digital signal. The peaks hold for a moment so that short transients can be monitored. **Each bar = 1dB.**

MAIN MIX · 0dB · -12dB · L R

Monitor Output Level

This control adjusts the monitor output level. Keep in mind that since the monitor level control comes after the Main Output Fader, nothing will be heard from your monitors if the main level is turned down.

Monitor Balance Control

This control sets the relative volume of the stereo monitor outputs and works just like the balance control on your home music system. This control is primarily used to make the volume from each speaker sound equal if you are not sitting exactly in the center of the two speakers.

Monitor Output Mute

This button completely cuts off the monitor output and provides a convenient way to instantly kill all sound without having to re-adjust the monitor level later. When the telephone rings, just hit the monitor mute to cut the noise.
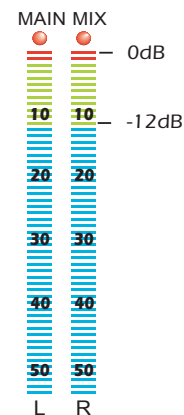
5 - Effects

Overview

PatchMix DSP comes complete with a host of great core DSP effects including Compressors, Delays, Choruses, Flangers and Reverb. Each 32-bit effect has various parameters for editing, as well as factory presets. You can also create and save as many of your own effect presets as you wish.

Since the effects are implemented in hardware, they don't place any load on your host computer. This allows your valuable CPU cycles to be used for other applications or software plug-ins. The effects are only available at the 44.1 and 48kHz sample rates.

There is a finite limit to how many effects you can use at the same time. As you use up the PatchMix DSP resources, certain effects will appear "grayed out" and cannot be added to the mixer. Complex effects such as reverb use more DSP resources than say a 1-Band EQ. If you continue to add effects, all of the DSP resources will eventually be used up.

❖ *Saving a session "defragments" the effect/ DSP resources. If you have used all your effects and need another, try saving the session.*

The Effects Palette

Click the FX button on the toolbar to bring up the Effects Palette. The Effects Palette contains two types of folders. The "Core Effects" folder contains the effect algorithms themselves. This folder cannot be modified. The other folders contain "Effects Chains", consisting of two or more effects grouped together. You can also add, delete, or modify Effects Chains and the folders that contain them. For more information on Effects Chains, see

New Folder Icon   Import/Export FX
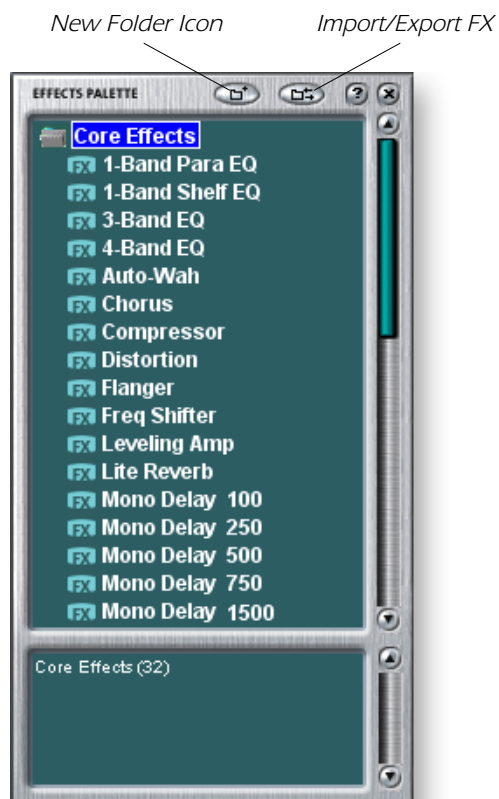
Effect Categories

Core Effects

Multi-Effects
Distortion Lo-fi
Drums & Percussion
Environment
Equalization
Guitar
Morpher
Multi Effects
Reverb
Synths & Keys
Vocal

EFFECTS PALETTE

Core Effects
- 1-Band Para EQ
- 1-Band Shelf EQ
- 3-Band EQ
- 4-Band EQ
- Auto-Wah
- Chorus
- Compressor
- Distortion
- Flanger
- Freq Shifter
- Leveling Amp
- Lite Reverb
- Mono Delay 100
- Mono Delay 250
- Mono Delay 500
- Mono Delay 750
- Mono Delay 1500

Core Effects (32)

▶ **To Select an Effect**

1. Click the FX button to bring up the Effects Palette. The effect palette contains numerous folders containing effects presets. Click on any folder to open it.

2. Select the effect you wish to use by clicking on it with the left mouse button and while continuing to hold the mouse button, drag the effect into the desired location on the PatchMix DSP mixer screen and release the mouse button. Multi effects contain several effects along with their parameter settings.

3. If you want to change the order of effects, simply Left-click and drag the effect to the desired location. Drag the effect to the area above or below the final destination and release the mouse button to move the effect.

❖ *The order of effects in a chain can have a big effect on the sound.*

This icon will appear when you drag an effect to a new location.

▶ **To Edit an Effect**

1. Click on the Insert Location containing the effect you wish to edit. The effect controls now appear on the TV screen.

2. Edit the effect parameters as desired.

▶ **To Delete an Effect**

1. Right-click on the Insert location containing the effect you wish to delete and a pop-up list appears.

2. Select "Delete Insert(s)" from the top of the list. The effect will be deleted.

FX Insert Chains

FX Insert Chains can be used to save several effects and their settings into a single multi-effect. When an effects chain is selected and dropped into an insert location, all the effects with control settings are copied as a single entity. Once dropped into an insert location, the effects are totally separate just as if you had placed them individually.

▶ **To Save FX Insert Chains**

1. Select two or more effects and place them into any consecutive insert locations.

2. Set the effect parameters the way you want them, including wet/dry mix settings.

3. Right-click to bring up the list of options.

4. Select "Save FX Insert Chain". The New FX preset dialog box appears.

5. Select a category folder where your preset will be placed, and enter a new preset name for your FX Chain.

❖ *Trim pots, peak meters and test tone generators will also be included in the FX chain.*

6. Select a folder where your new preset will be placed, then type in a new preset name and click **OK**. Your preset is now saved.

Creating, Renaming & Deleting Categories or Presets
There are several utilities to help you organize your effects presets.

▶ To Create a New Preset Category
You can create your own category folders to help organize your effects presets.

1. Left-click on the New Folder icon at the top of the Effects Palette. A pop-up dialog box appears asking you to "Enter the Name of the New Category."

   • Alternatively, you can Right-click over an Effects Folder, which calls a pop-up dialog box with the option to "Create New Category."

2. Type in a name for your new folder.

3. Click **OK** to create a new folder or Cancel to cancel the operation.

▶ To Delete an Effect Category or Preset
1. Right-click on the category folder you wish to delete. A pop-up selection box appears.

2. Select "Delete Category". A popup dialog box appears warning you that this action will delete all presets in the folder.

3. Click **OK** to delete the folder or Cancel to cancel the operation.

▶ To Rename an Effects Category
1. Right-click on the category folder you wish to rename. A pop-up selection box appears.

2. Select "Rename Category". A pop-up dialog box appears, asking you to "Enter New Category Name."

3. Click **OK** to rename the folder or Cancel to cancel the operation.

Importing and Exporting Core FX Presets and FX Insert Chains

These utilities make it easy to import or export your FX Presets and FX Insert Chains. You can share presets with your friends or download new presets from the Internet.

▶ **To Import Core FX Presets**

This option imports complete folders of Core FX presets into the E-MU PatchMix DSP folder (normally located here: "C:\Program Files\Creative Professional\E-MU PatchMix DSP\Core Effects"). If the name of an imported FX preset exactly matches a preset you already have, a number will be appended to end of the imported preset name.

1. Click the **Import/Export FX Library** button from the FX Palette.
2. Select **Import FX Library**. The "Browse for Folder" window appears.
3. **Choose the folder** where the Core FX presets you wish to import are located.
4. The selected folder of Core FX presets will be copied into the **Core Effects** folder of PatchMix DSP.

▶ **To Import FX Category Folders**

This option imports complete category folders of FX Chains into the E-MU PatchMix DSP folder (normally located here: "C:\Program Files\Creative Professional\E-MU PatchMix DSP\Effect Presets"). If the name of an imported FX preset exactly matches a preset you already have, a number will be appended to end of the imported preset name.

1. Click the **Import/Export FX Library** button from the FX Palette.
2. Select **Import FX Category**. The "Browse for Folder" window appears.
3. **Choose the folder** where the FX Chains you wish to import are located.
4. The selected folder of FX Chains will be copied into the **Effect Presets** folder of PatchMix DSP.

▶ **To Export your Core FX Presets**

This option exports your Core FX presets to a folder of your choice.

1. Click the **Import/Export FX Library** button from the FX Palette.
2. Select **Export FX Library**. The "Browse for Folder" window appears.
3. Choose a destination location for the Core FX presets, then press **OK**.
4. The Core FX presets will be copied to the selected destination.

▶ **To Export your FX Category Folders**

This option exports a single category of FX chains to a folder of your choice.

1. Click the **Import/Export FX Library** button from the FX Palette.
2. Select **Export FX Category**. A pop-up dialog box appears asking you to "Choose the FX Category to be exported".
3. **Choose the desired FX Category** to export. Press **OK** to continue or **Cancel** to cancel the operation.
4. The "Browse for Folder" window appears. **Choose a destination** location for the Core FX presets, then press **OK**.
5. The FX Chains will be copied to the selected destination.

FX Edit Screen

Click on an FX Insert to display the parameters for that effect. If an insert effect is not selected, the FX display will read "No Insert".

Most effects have a wet/dry mix parameter to control the ratio of effect-to-plain signal. The wet/dry setting is stored with the FX preset. The effect parameters vary with the type of effect. Generally if an effect is placed in an Aux Send, the wet/dry mix in the effect should be set to 100% wet since the Aux Return amount controls how much effect is applied.

The User Preset section is located at the bottom of the FX Edit screen. User presets are variations of the main effect and can be edited, deleted, renamed or overwritten as you wish.

▪▪ Note: *Effects have to be placed into an insert location before you can program them.*
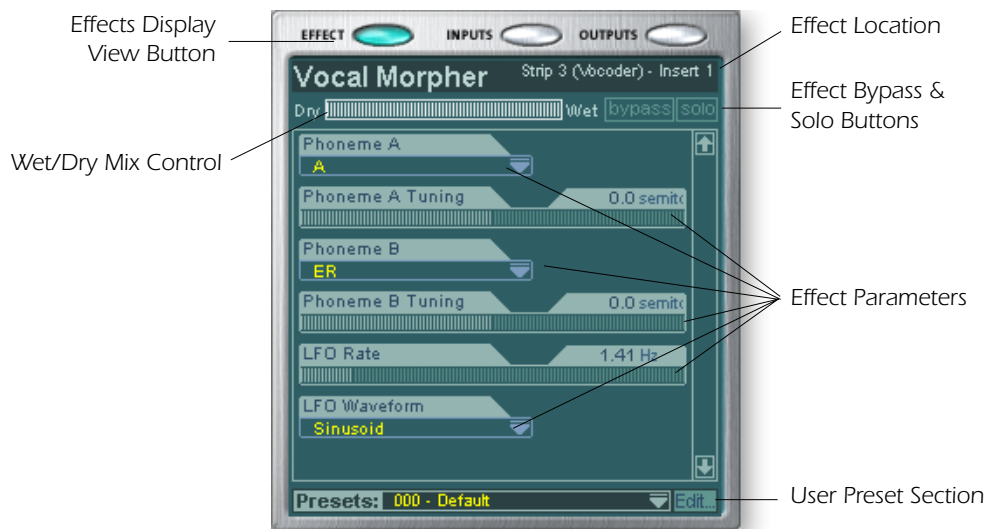
▶ To Bypass an Insert:

Inserts can be bypassed if you want to temporarily hear the audio without the effect or insert. Bypass can also be used to turn off a Send Insert.

Method #1

1. Click on the Effect (in the Insert section)
2. Click the Bypass button in the TV display.

Method #2

1. Right-click over the Insert you want to bypass (in the Insert section). A pop-up menu appears.
2. Select "Bypass Insert" from the list of options. The insert effect name will "gray-out" to indicate that the insert effect is bypassed.

▶ To Solo an Insert:

Inserts can also be soloed. Solo bypasses all the other inserts in the strip and allows you to hear only the soloed effect. This feature is very useful when adjusting the effect parameters.

Method #1

1. Click on the Insert Effect (in the Insert section).
2. Click the Solo button in the TV display.

Method #2

1. Right-click over the Insert Effect you want to Solo (in the Insert section). A pop-up menu appears.

2. Select "Solo Insert" from the list of options. The other Insert Effect names in the strip will "gray-out" to indicate that they are bypassed.

▶ **To Bypass ALL**

All the inserts in a strip can be bypassed with a single command.

1. Right-click over any Effect in the Insert section. A pop-up menu appears.

2. Select "Bypass All Inserts" from the list of options. All the insert names will be "grayed-out" to indicate that they are bypassed.

▶ **To Un-Bypass ALL**

All the inserts in a strip can also be un-bypassed with a single command. This command works even if only some of the effects are bypassed.

1. Right-click over any Effect in the Insert section. A pop-up menu appears.

2. Select "Un-Bypass All Inserts" from the list of options. All the insert names will light to indicate that they are active.

User Preset Section

Each core effect has a set of User Presets, that you can use to store your favorite effect parameter settings. We've included a good collection of user presets to get you started. The user presets are accessed from the bar at the bottom of the TV screen. The user preset edit menu allows you to select stored presets, create new presets, rename or delete existing presets, or overwrite existing presets with your modified settings. User presets stay with the Mixer application regardless of which Session is open.

▪▪ To copy or share User Presets, you must save them as FX Palette effects.

Presets: 002 - Jet Flanger [user] ▼ Edit... ——— *Click here for Edit Menu*

Click here to Select Presets

▶ **To Select a User Preset**

1. Select the FX display in the TV screen.

2. Select the desired insert effect, highlighting it. The effect parameters appear in the TV screen.

3. Click on the ▼ icon on the preset menu. A drop-down preset list appears.

4. Select a preset from the list.

▶ **To Create a New User Preset**

1. Select the FX display in the TV screen.

2. Select the desired insert effect, highlighting it. The effect parameters appear in the TV screen.

3. Click on the Edit button. A pop-up menu appears.

4. Select New. A pop-up dialog box appears asking you to name the new preset.

5. Name the preset and click OK. Your new preset is now saved.

▶ **To Delete a User Preset**

1. Select the user preset you wish to delete from the user preset menu.

2. Click on the Edit button. A pop-up menu appears.

3. Select Delete. A pop-up dialog box appears asking you to confirm your action.

4. Click OK to delete the preset or No or Cancel to cancel the operation.

▶ **To Rename a User Preset**

1. Select the user preset you wish to rename from the user preset menu.

2. Click on the Edit button. A pop-up menu appears.

3. Select Rename. A pop-up dialog box appears asking you to rename the preset.

4. Type in the new preset name, then click OK to rename the preset or Cancel to cancel the operation.

▶ **To Overwrite or Save a User Preset**

This operation allows you to overwrite an existing preset with a newer version.

1. Select the user preset you wish to modify from the user preset menu and make any changes you wish.

2. Click on the Edit button. A pop-up menu appears.

3. Select Overwrite/Save. The current preset will be overwritten with the new settings.

Core Effects and Effects Presets

The Core Effects cannot be removed or copied. Effect presets (stored in "C:\Program Files\Creative Professional\Digital Audio System\E-MU PatchMix DSP\Effect Presets") can be copied, e-mailed or shared like any other computer file.

Hint: You can open the effects presets with "NotePad" or other word processor to view and edit the name and parameters.

List of Core Effects

Stereo Reverb	Frequency Shifter	Mono Delay 750
Lite Reverb	Auto-Wah	Mono Delay 1500
Compressor	Vocal Morpher	Mono Delay 3000
Leveling Amp	1-Band Para EQ	Stereo Delay 100
Chorus	1-Band Shelf EQ	Stereo Delay 250
Flanger	3-Band EQ	Stereo Delay 500
Distortion	4-Band EQ	Stereo Delay 750
Speaker Sim	Mono Delay 100	Stereo Delay 1500
Rotary	Mono Delay 250	
Phase Shifter	Mono Delay 500	

DSP Resource Usage

There are two main factors which determine the total number of effects available for use at any given time: Tank Memory and DSP Instructions. Using too much of either resource will cause effects to be unavailable (grayed out) in the FX menu. In addition, the strips themselves use DSP Instructions, so only create strips that you actually need.

Tank memory is the memory used by delay-based effects such as reverb and digital delays. All the reverbs and delays aside from the Mono Delay 100 and Stereo Delay 100 use varying amounts of tank memory.

The DSP instructions are used by all the effects. Effects with multiple stages, such as multi-band EQs or the speaker simulator use more DSP instructions than a 1-Band EQ.

Tank memory tends to get used first, and so we've provided many delay line effects to allow maximum conservation of this precious resource. Use only the longest delay you actually need.

The chart below shows three possible effects combinations. These were created by using up the reverb resources first. Even more simultaneous effects are possible if fewer reverbs and shorter delays are used.

❖ **Tip:** Saving a session "defragments" the effect/ DSP resources. If you have used all your effects and need another, try saving the session.

Examples of Effects Usage *(with a WAVE, ASIO Return & 2 Inputs)*

Example 1	No.	Example 2	No.	Example 3	No.
Stereo Reverb	2	Lite Reverb	5	Stereo Reverb	1
4-Band EQ	4	3-Band EQ	5	Lite Reverb	2
3-Band EQ	2	1-Band EQ	4	Stereo Delay 1500	1
1-Band EQ	6	Compressor	1	Mono Delay 250	1
Compressor	6	Mono Delay 1500	1	Compressor	6
Chorus	1	Mono Delay 250	1	Chorus	2
Mono Delay 1500	1	Auto-Wah	1	Flanger	2
				4-Band EQ	3
				3-Band EQ	3
Total Effects	22	**Total Effects**	18	**Total Effects**	21

Core Effects Descriptions

1-Band Para EQ
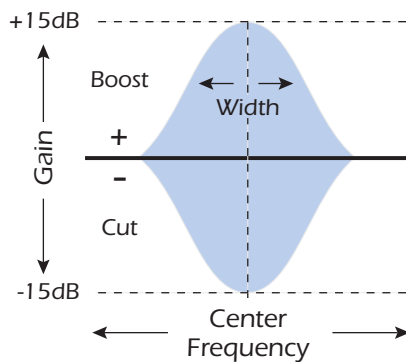
This single band parametric equalizer is useful when you just want to boost or cut a single range of frequencies. For example, if you just want to brighten up the lead vocal a bit, you might choose this EQ. This EQ offers up to ±15dB cut or boost.

| Parameter	Description
Gain	Sets the amount of cut (-) or boost (+) of the selected frequency band. Range: -15dB to +15dB
Center Frequency	Sets the range of frequencies to be cut or boosted with the Gain control. Range: 80Hz 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: 1semitone to 36 semitones

1-Band Shelf EQ

This single band shelving equalizer is useful when you just want to boost or cut a single range of frequencies at the high or low end of the spectrum. For example, if you just want to add a little more bass, there's no need to waste a 3-band EQ. Just choose low shelf, then adjust the gain and frequency. This EQ offers up to ±15dB cut or boost.
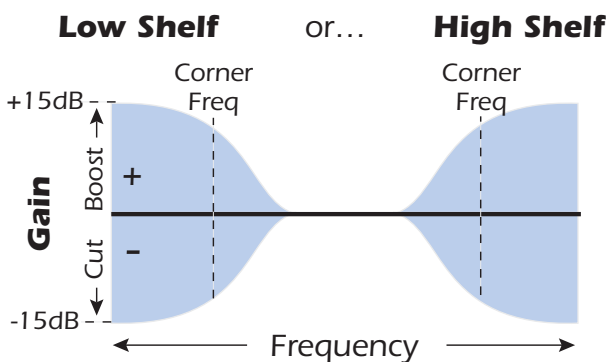
Parameter	Description
Shelf Type	Allows you to choose either low shelving or high shelving EQ.
Gain	Sets the amount of cut (-) or boost (+) of the shelf. Range: -15dB to +15dB
Corner Frequency	Sets the frequency where the signal begins getting cut or boosted with the Gain control. Range: 80Hz to 16kHz

3-Band EQ

This versatile equalizer provides two shelving filters at the high and low ends of the frequency range and a fully parametric band in the center. Up to ±24 dB of boost or cut is provided for each band.
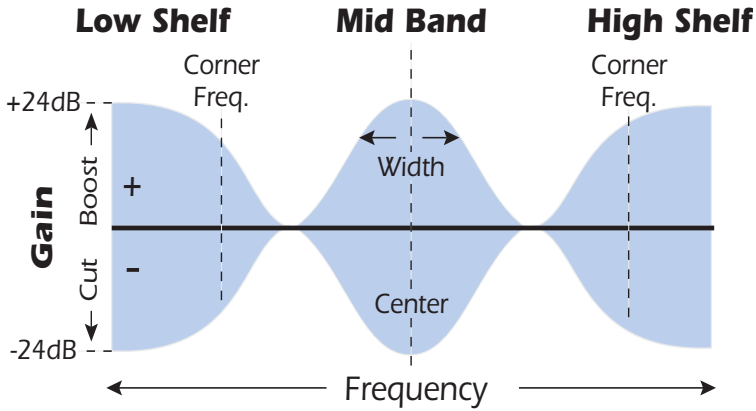
> ▪▪ **Note:** The Wet/Dry Mix control on an equalizer should normally be set to 100% wet or unpredictable results may occur.

▶ **Setting up a Parametric EQ**

1. Turn up the gain on the band you are working with. This allows you to easily hear the effect of the filter.

2. Reduce the bandwidth if you are working with a mid-band.

3. Adjust the Center Frequency to "zero-in" on the frequencies you wish to boost/cut.

4. Set the Gain to a positive value to boost frequencies or to a negative value to cut out frequencies.

5. Widen the Bandwidth to create a more natural sound.

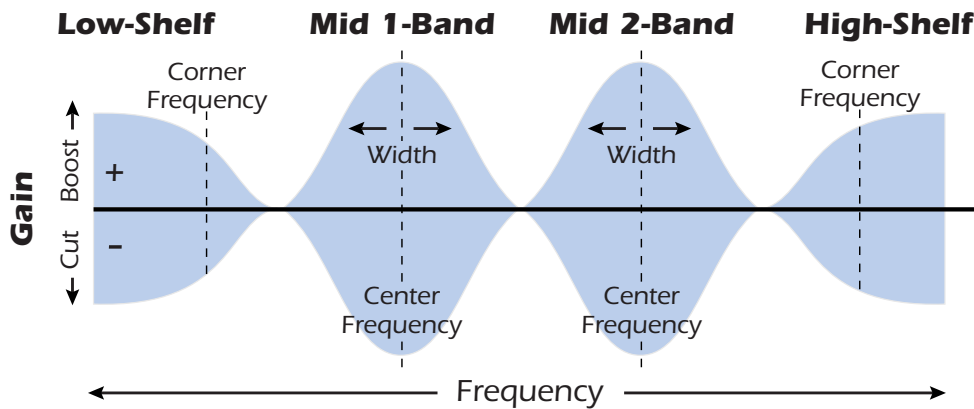6. Adjust and tweak as needed.

Parameter	Description
High Shelf Gain	Sets the amount of cut (-) or boost (+) of the high frequency shelf. Range: -24dB to +24dB
High Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the High Gain control. Range: 4kHz to 16kHz
Mid Gain	Sets the amount of cut (-) or boost (+) of the mid frequency band. Range: -24dB to +24dB
Mid Freq. 1	Sets the range of frequencies to be cut or boosted with the Mid Gain control. Range: 200Hz to 3kHz
Mid Bandwidth	Sets the width of the frequency range for the Mid Center Frequency band that will be cut or boosted by the Mid Gain control. Range: 1 semitone to 1 octave
Low Shelf Gain	Sets the amount of cut (-) or boost (+) of the low frequency shelf. Range: -24dB to +24dB
Low Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the Low Gain control. Range: 50Hz to 800Hz

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 ±24 dB of boost or cut is provided for each band.

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

For more information about setting up a parametric EQ, see .

Parameter	Description
High Shelf Gain	Sets the amount of cut (-) or boost (+) of the high frequency shelf. *Range: -24dB to +24dB*
High Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the High Gain control. *Range: 4kHz to 16kHz*
Mid 2 Gain	Sets the amount of cut (-) or boost (+) of the Mid 2 Frequency band. *Range: -24dB to +24dB*
Mid 2 Center Freq.	Sets the range of frequencies to be cut or boosted with the Mid 2 Gain control. *Range: 1kHz to 8kHz*
Mid 2 Bandwidth	Sets the width of the frequency range for the Mid 2 Center Frequency band that will be cut or boosted by the Mid 2 Gain control. *Range: .01 octave to 1 octave*
Mid 1 Gain	Sets the amount of cut (-) or boost (+) of the Mid 1 Frequency band. *Range: -24dB to +24dB*
Mid 1 Center Freq.	Sets the range of frequencies to be cut or boosted with the Mid 1 Gain control. *Range: 200Hz to 3kHz*
Mid 1 Bandwidth	Sets the width of the frequency range for the Mid 1 Center Frequency band that will be cut or boosted by the Mid 1 Gain control. *Range: .01 octave to 1 octave*
Low Shelf Gain	Sets the amount of cut (-) or boost (+) of the low frequency shelf. *Range: -24dB to +24dB*
Low Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the Low Gain control. *Range: 50Hz to 800Hz*

Auto-Wah

This effect creates the sound of a guitar wah-wah pedal. The "Wah" filter sweep is automatically triggered from the amplitude envelope of the input sound. Auto-wah works well with percussive sounds such as guitar or bass.

The Auto-Wah is a bandpass filter whose frequency can be swept up or down by an envelope follower, which extracts the volume contour of the input signal. The Envelope Sensitivity setting allows you to properly set up the envelope follower to receive a wide variety of input signals. This "envelope", or volume contour, controls the frequency of the bandpass filter so that it sweeps up and down with each new note. The Attack controls the rate of the note-on sweep. As the input sound fades away, the filter sweeps back at a rate determined by the Release setting.

The wah direction allows the filter to be swept either up or down in frequency. Use a higher Center Frequency setting when the wah direction is down.

Auto-Wah Filter
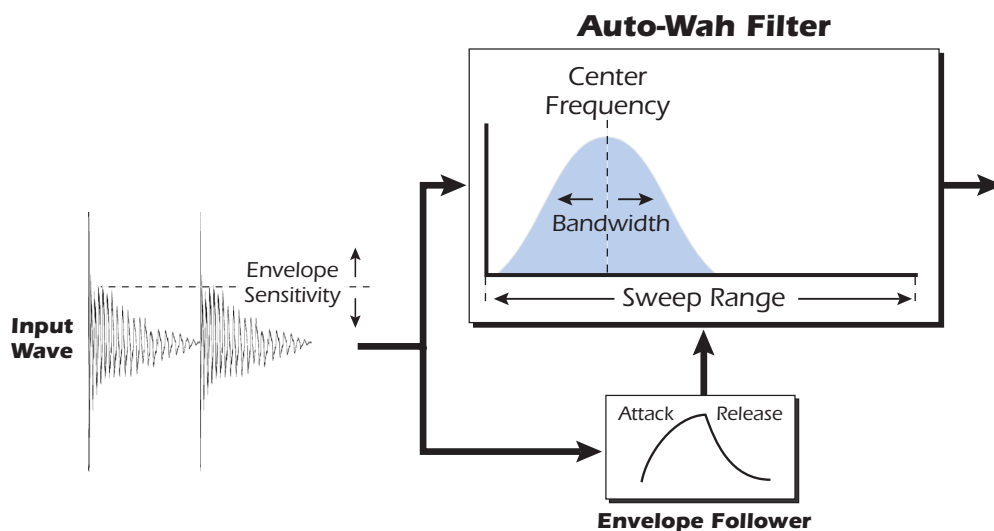
| Parameter	Description
Wah Direction	Allows you to sweep the wah up or down.
Env. Sensitivity	Controls how closely the wah sweep follows the input signal. Range: -12dB to +18dB
Env. Attack Time	Sets the starting rate of the "wah" sweep. Range: 0ms to 500ms
Env. Release Time	Sets the ending or release rate of the "wah" sweep. Range: 10ms to 1000ms
Sweep Range	Controls the amount of "wah" sweep. Range: 0% to 100%
Center Frequency	Sets the initial bandpass filter frequency. Range: 80Hz to 2400Hz
Bandwidth	Sets the width of the bandpass filter. Range: 1Hz to 800Hz

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. 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). Since this is a stereo chorus, an LFO phase parameter is included which can be used to widen the stereo image.

Parameter	Description
Delay	Sets the length of the delay. Range: 0ms to 20ms.
Feedback	Sets the amount of delayed signal that will be recirculated through the delay line. Range: 0% to 100%
LFO Rate	Sets the frequency of the low frequency oscillator. Range: .01Hz to 10Hz
LFO Depth	Sets how much the LFO affects the delay time. Increases the animation and amount of the chorus effect. Range: 0% to 100%
LFO Waveform	Selectable between Sine or Triangle wave.
LFO L/R Phase	Controls the stereo width by adjusting the phase difference of the LFO waveform between left and right channels. Range: -180° to +180°

Compressor

In its simplest form, an audio compressor is just an automatic gain control. When the volume gets too loud, the compressor automatically turns it down. Compressors are useful in musical applications because they allow you to record a "hotter" signal without overloading the recording device.

Since the compressor turns **down** the gain of the signal, you might wonder how can it make the signal level stronger. A Post Gain control allows you to boost the output gain of the compressor in order to make up for the gain reduction. The overall level is higher and only turned down when the signal level gets too loud. This level is called the *Threshold*, which just happens to be the most important control on the compressor.
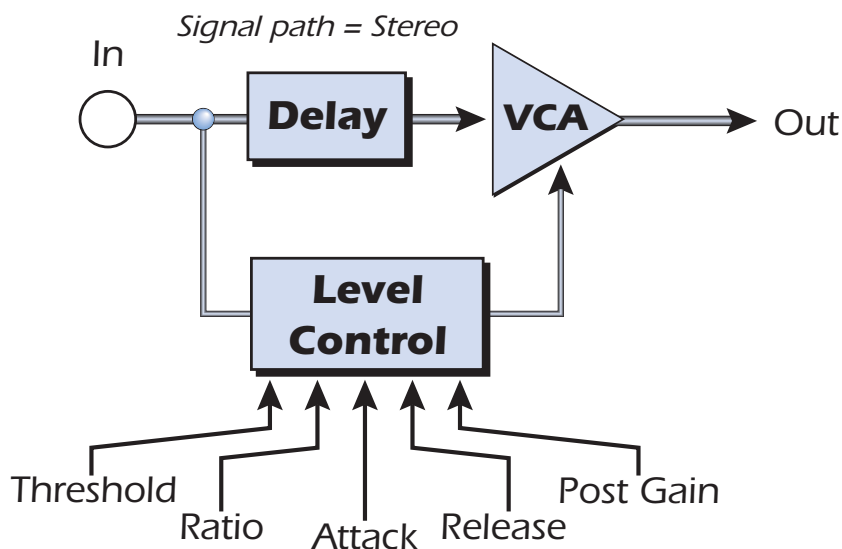
Basic Controls

The three main controls of a compressor are the *Ratio* control, the *Threshold* control and the *Gain* control.

If the signal falls below the **Threshold**, no processing will take place. Signals exceeding the Threshold will have gain reduction applied as set by the ratio control. This important control allows you to dial in the range of amplitudes you want to tame. For example, if you're trying to trim off just the loudest peaks, set the threshold so the gain reduction meter only shows compression during these peaks. One of the biggest mistakes in using a compressor is having the threshold set too low. This adds noise as the compressor will always be reducing the volume.

The **Ratio** control determines how strongly the compressor will affect the signal. The higher the ratio, the more reduction will be applied. If the ratio is high enough, *(above 10:1)* the signal will effectively be prevented from getting any louder. In this situation, the compressor will be acting as a **Limiter**, placing an upper limit on the signal level. In general, ratios from 2:1 to 6:1 are considered compression and higher ratios above 10:1 are considered limiting.

The **Post Gain** control amplifies the signal after it has been compressed to bring it back up in volume. If you don't increase the gain, the compressed signal will be much lower in volume.

Two other important controls are **Attack** and **Release**. Attack controls how quickly the gain is turned down after the signal exceeds the threshold. Release controls how fast the gain is returned to its normal setting after the signal has fallen below the threshold again. An attack setting of about 10 milliseconds will delay the onset of compression long enough to preserve the attack transients in guitar, bass or drums while allowing the sustain portion of the sound to be compressed. Longer release times are generally used to reduce the so called "pumping" effect as the compressor turns on and off. Don't make the release time too long, however, or the compressor won't have time to recover for the next pluck or hit. In general, the attack and release controls are used to smooth out the action of the compressor, but they can also be used to create special effects.

The Pre-Delay parameter lets the level detector "look into the future" up to 4 milliseconds in order to anticipate upcoming peaks in the signal. This is accomplished of course, by inserting delay into the signal path. This lookahead technique allows the use of slower attack times without missing signal peaks. This parameter is especially effective on drums and percussion.

The Input Meter allows you to monitor the strength of your input signal. Always try to boost the signal before the compressor if you can.

The Compression Meter shows the amount of gain reduction being applied. Since this meter displays how much the gain is being turned *down*, the meter moves from right to left, instead of left to right like a normal meter.

Parameter	Description
Threshold	Threshold sets the input signal level above which dynamic range compression takes place. Everything above the threshold will be brought down in volume. Range: -60dB to +12dB
Ratio	Sets the ratio of input signal level to output signal level, or "how much" compression will be applied. Range: 1:1 to ∞:1
Post Gain	Amplifies the signal after it has been compressed to bring up the volume. Range -60dB to +60dB

Parameter	Description
Attack Time	Controls how quickly the gain is turned down after the signal exceeds the threshold. Range .1ms to 500ms
Release Time	Controls how fast the gain is returned to its normal setting after the signal has fallen below the threshold. Range: 50ms to 3000ms
Pre-Delay	Allows the use of slower attack times without missing signal peaks. Range: 0ms to 3 ms

**Input Meter** Allows you to monitor the strength of the input signal.

Gain Reduction Meter Shows the amount of gain reduction being applied.

Distortion

Most audio processors aim to provide low distortion, but not this one! The sole purpose of this effect is to add distortion, and lots of it. This effect provides "fuzz box" style, clipping distortion which is particularly effective on guitar, bass, organs, electric pianos or whatever.

The input signal first passes through a lowpass filter. The Lowpass Filter Cutoff Frequency allows you to control the number of new harmonics that will be generated by the distortion element. The distortion element has an Edge control which controls "how much" distortion will be added. A bandpass filter follows the distortion generator. The EQ Center control lets you select a particular band of frequencies to be output. The EQ Bandwidth controls the width of the center frequency band. Finally, a gain control allows you to make up for any gain loss through the effect.

Use the Wet/Dry mix control in conjunction with the Edge control to reduce the amount of distortion, or go wild and turn everything to 11!
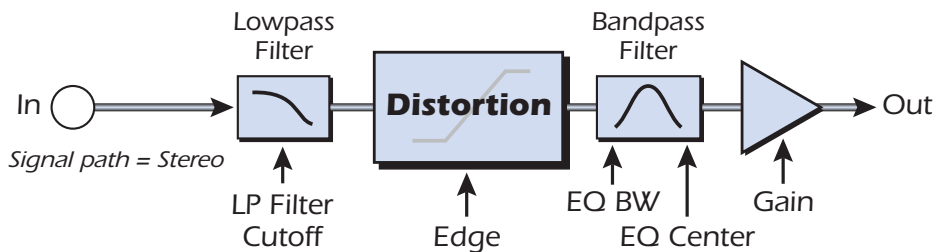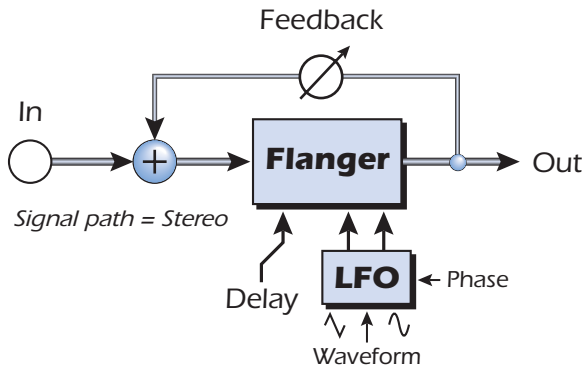

Parameter	Description
Pre EQ LP Cutoff	Controls the amount of high frequency audio admitted to the distortion. Range: 80Hz to 24kHz
Edge	Sets the amount of distortion and new harmonics generated. Range: 0-100
Gain	Sets the output volume of the effect. Range: -60dB to 0dB
Post EQ Center Freq.	Sets the frequency of the output bandpass filter. Range: 80Hz to 24kHz
Post EQ Bandwidth	Sets the width of the output bandpass filter. Range: 80Hz to 24kHz

Flanger

A flanger is a very short delay line whose output is mixed back together with the original sound. Mixing the original and delayed signals results in multiple frequency cancellations known as a comb filter. Since the flanger is a type of filter, it works best with harmonically rich sounds.
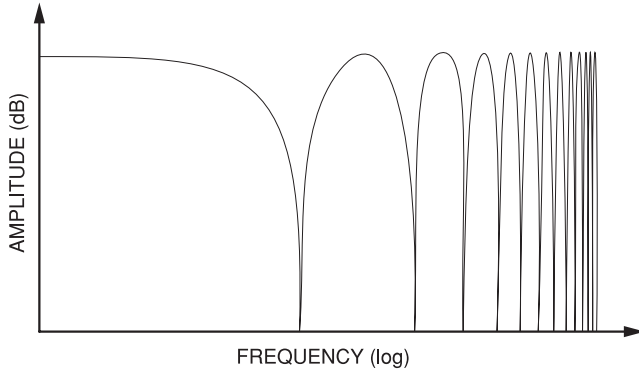
A low frequency oscillator is included to slowly change the delay time. This creates a rich, sweeping effect as the notches move up and down across the frequency range. The amount of feedback deepens the notches, intensifying the effect. You can invert the feedback signal by choosing a negative feedback value. Inverting the feedback signal creates peaks in the notch filter and deepens the effect.

Parameter	Description
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
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%
LFO Rate	Sets the speed of the flanger sweep. Range: .01 Hz to 10Hz
LFO Depth	Sets how much the LFO affects the delay time. Increases the animation and amount of the flanging effect. Range 05 to 100%
LFO Waveform	Selectable between Sine or Triangle wave.
LFO L/R Phase	Controls the stereo width by adjusting the phase difference between the left and right sweeps. Range: -180° to +180°

Freq Shifter

This unusual effect is sometimes called "spectrum shifting" or "single sideband modulation". Frequency shifting shifts every frequency in the signal by a fixed number of Hz which causes the harmonics to lose their normal relationship. The more common pitch shifter, in contrast, preserves the harmonic relationships of the signal and so is better suited to creating "musical" harmonies.

This isn't to say that the frequency shifter can't be used musically. Small intervals of frequency shifting (1 Hz and below) can produce a wonderful, lush chorusing or phasing effect. For bizarre frequency shifting effects, simply crank up the frequency knob. Frequencies can be shifted up or down by any specified amount from .1 Hz to 24 kHz. You can also shift pitch up on one side and down on the other if you wish.

❖ You can also type in exact frequencies to a resolution of 1/10 Hz.

Comparison between Pitch and Frequency Shifting

Harmonic	Original (Hz)	Pitch Shifted (100 Hz)	Frequency Shifted (100 Hz)
1	200	300	300
2	400	600	500
3	600	900	700
4	800	1200	900
5	1000	1500	1100
6	1200	1800	1300
7	1400	2100	1500
8	1600	2400	1700

Parameter	Description
Frequency	Sets the number of Hz that will be added or subtracted with every harmonic in the signal. Range: .01Hz to 24kHz
Left Direction	Sets pitch shift up or down for the left channel.
Right Direction	Sets pitch shift up or down for the right channel.

Leveling Amp

The first compressors developed in the 1950's were based on a slow-acting optical gain cells which were able to control the signal level in a very subtle and musical way. This effect is a digital recreation of the leveling amps of yesteryear.

The leveling amp uses a large amount of "lookahead delay" to apply gentle gain reduction. Because of this delay, the leveling amp is not suitable for applications which require realtime monitoring of the signal. This smooth and gentle compressor is designed to be used in situations where delay does not pose a problem, such as mastering a mix or compressing prerecorded stereo material.

Post Gain is the only control on the leveling amp. This control is used to make up the volume lost by the compression. The Compression Ratio is fixed at about 2.5:1. If a large peak is detected, the effect will automatically increase the compression ratio to keep the audio output controlled.

The gain reduction meter shows you how much gain reduction is being applied. Since the gain reduction meter displays how much the gain is being turned *down*, the meter moves from right to left, instead of left to right like most meters.

Post Gain	Amplifies the signal after it has been compressed to bring up the volume. Range 0dB to 36dB

Lite Reverb

Reverberation is a simulation of a natural space such as a room or hall. The Lite Reverb algorithm is designed to simulate various rooms and reverberation plates while using fewer DSP resources than the Stereo Reverb. Up to five Lite Reverbs can be used at once.

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.

Early Reflections　　**Reverberation**　　Time
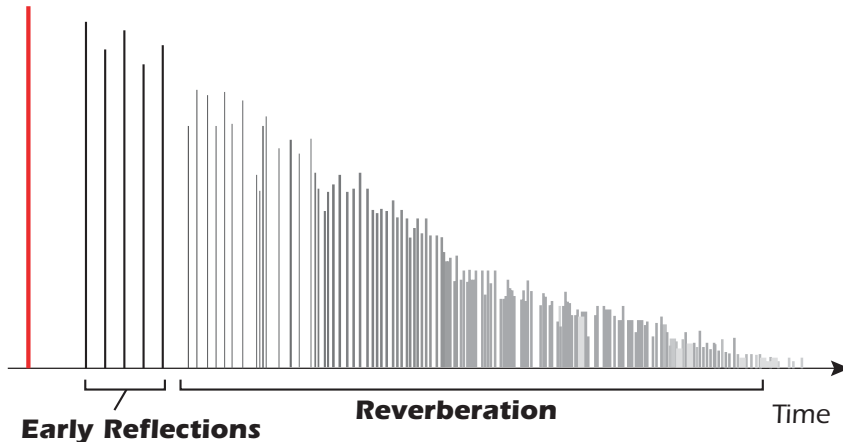
After a short pre-delay period, the echoes from the closest walls or ceiling are heard. These first echoes, or *Early Reflections*, vary greatly depending on the type of room. Some time after the early reflection cluster ends, the actual *Reverberation* (a dense cloud of complex wall reflections) begins and decays according to the time set by the Decay Time parameter. The Reverberance parameter controls the density and smearing of both the early reflections and the reverberation cloud.

High frequency energy tends to fade away first as a sound is dissipated in a room. The High Frequency Decay Factor 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.

The Low Frequency Decay Factor parameter adjusts the time it takes for the low frequencies to die away. This control adjusts the "boominess" of the room.

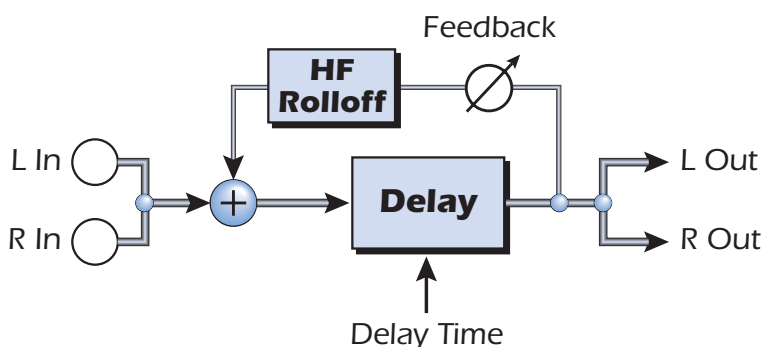| Parameter	Description
Decay Time	Sets the reverb decay time. Range: 0% to 100%
HF Decay Factor	Sets the rate at which high frequencies die away. The high frequencies last longer as the percentage is increased. Range: 0% to 100%
LF Decay Factor	Sets the rate at which low frequencies die away. The low frequencies last longer as the percentage is increased. Range: 0% to 100%
Early Reflections	Sets the volume of the initial wall reflections. Range: 0% to 100%
Reverberance	Sets the amount of scattering of the early reflections and the reverberation cloud. Range: 0% to 100%

Mono Delays - 100, 250, 500, 750, 1500, 3000

A delay line makes a copy of the incoming audio, holds it in memory, then plays it back after a predetermined time. The delay number refers to the maximum delay time that can be produced by the delay line. The six lengths, from 100 ms to 3 seconds, allow you to make the most efficient use of the effect memory resource.

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 the Mono Delay.

There is also a feedback path to send the delayed audio back through the delay line. When creating echo effects, the feedback controls how many echoes will be produced. With short delays, the feedback control acts as a *resonance* control, increasing the amount of comb filtering produced by the delay line.

A High Frequency Rolloff filter 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.

Parameter	Description
Delay Time	Sets the length of the delay in milliseconds. *(.01ms. minimum increment between settings)*
Mono Delay 100	Range: 1 millisecond to 100 milliseconds
Mono Delay 250	Range: 1 millisecond to 250 milliseconds
Mono Delay 500	Range: 1 millisecond to 500 milliseconds
Mono Delay 750	Range: 1 millisecond to 750 milliseconds
Mono Delay 1500	Range: 1 millisecond to 1.5 seconds
Mono Delay 3000	Range: 1 millisecond to 3 seconds
Feedback	Sets the amount of delayed signal that will be recirculated through the delay line. Range: 0% to 100%
High Freq. Rolloff	Damps high frequencies in the feedback path. Range: 0% to 100%

Phase Shifter

A 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.

By setting the LFO Depth to zero and tuning the LFO Center, a fixed multi-notch filter is created.
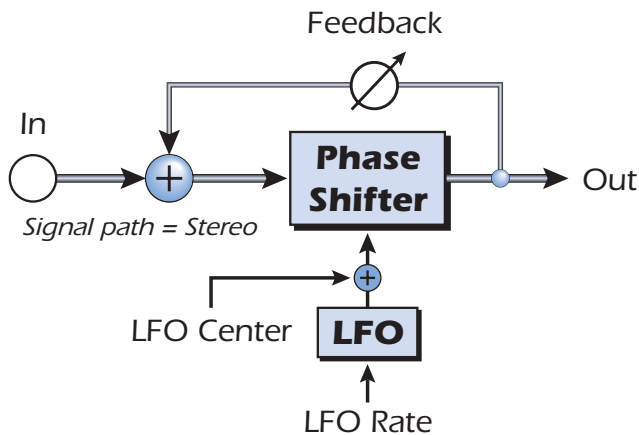
Parameter	Description
LFO Center	Sets the initial offset of the LFO and changes the position of the peaks and notches. Range: 0% to 100%
Feedback	Increases the depth of the notches and height of the peaks. Range: 0% to 100%
LFO Rate	Controls the sweep rate of the Low Frequency Oscillator. Range: .01Hz to 10Hz
LFO Depth	Controls how much the Center Frequency is swept by the LFO. Range: 0% to 100%
Waveform	Selects a Sine or Triangle wave for the LFO
LFO L/R Phase	Controls the stereo width by adjusting the phase difference between the left and right sweeps. Range: -180° to +180°

Rotary

This is a simulation of a rotating speaker used on organs. The rotating speaker was invented to give static organ tones a pipe organ type of animation, but this distinctive sound became a legend in its own right. Spinning a sound around the room creates a doppler pitch shift along with many other complex and musically pleasing sonic effects.

The Rotary incorporates acceleration and deceleration as you switch between the two speeds.

Parameter	Description
Speed	Switches between slow or fast rotor speeds with acceleration and deceleration as the speed changes.

Speaker Simulator

The Speaker Simulator provides realistic guitar speaker responses and is designed for use with guitar, bass or synthesizer. Twelve popular guitar amp speaker cabinets are modeled.

There is only one parameter on this effect. Just select the speaker you want and listen. Normally this effect should be used with the Mix control set to 100%.

Speaker Type	Description
British Stack 1 & 2	Modeled from a British 8-speaker high power amplifier stack.
British Combo 1-3	Modeled from a British 2-speaker combo amplifier.
Tweed Combo 1-3	Modeled from an American, 1950's era, 2-speaker combo amplifier.
2 x 12 Combo	Modeled from an American, 1960's era, 2-speaker combo amplifier.
4 x 12 Combo	Modeled from an American, 1960's era, 4-speaker amplifier set.
Metal Stack 1 & 2	Modeled from a modern era, power amplifier stack.

Stereo Delays - 100, 250, 500, 750, 1500

The Stereo Delays are true stereo delay lines in that the left and right channels are kept entirely separate from each other. The delay number refers to the maximum delay time that can be produced by the delay lines. The five different lengths, from 100 ms to 1.5 seconds, allow you to make the most efficient use of the effect memory resource.

Because the left and right channels can have different delay times, you can create a panning effect by setting one delay long and the other short. Very short delay times combined with a high feedback amount can be used to create monotone robotic-sounding effects. Using the longer stereo delays, you can "overdub" musical lines one on top of the other with the feedback control turned up.
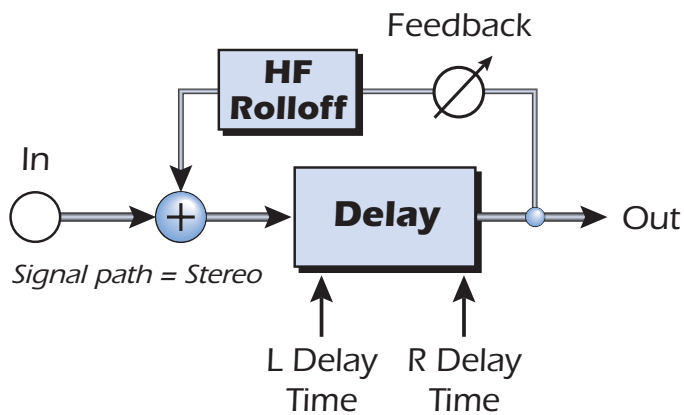
Parameter	Description
Left Delay Time	Sets the length of the delay for the left channel in milliseconds.
Right Delay Time	Sets the length of the delay for the right channel in milliseconds.
Delay Time (L & R) Stereo Delay 100 Stereo Delay 250 Stereo Delay 500 Stereo Delay 750 Stereo Delay 1500	*(.01ms. minimum increment between settings)* Range: 1 millisecond to 100 milliseconds Range: 1 millisecond to 250 milliseconds Range: 1 millisecond to 500 milliseconds Range: 1 millisecond to 750 milliseconds Range: 1 millisecond to 1.5 seconds
Feedback	Sets the amount of delayed signal that will be recirculated through the delay line. Range: 0% to 100%
High Freq. Rolloff	Damps high frequencies in the feedback path. Range: 0% to 100%

Stereo Reverb

Reverberation is a simulation of a natural space such as a room or hall. The stereo reverb algorithm is designed to simulate various halls, rooms and reverberation plates.

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.
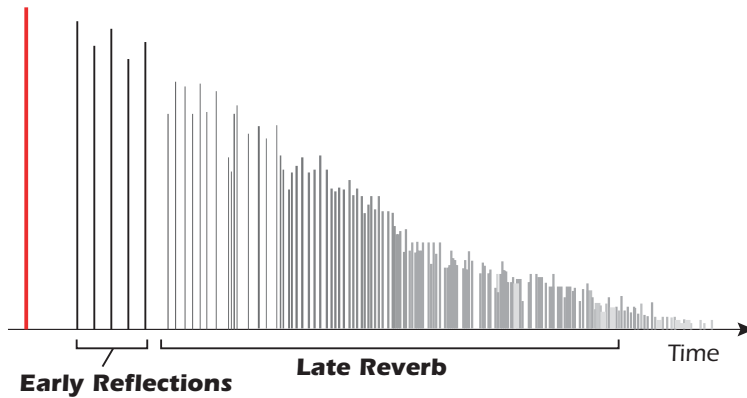
After a short pre-delay period, the echoes from the closest walls or ceiling are heard. These first echoes, or early reflections, vary greatly depending on the type of room. Some time after the early reflection cluster ends (late reverb delay), the late reverberation (a dense cloud of complex wall reflections) begins and decays according to the time set by the Decay Time parameter.

Diffusion is the amount of scattering and density of the late 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.

The Low Frequency Damping parameter adjusts the time it takes for the low frequencies to die away. This control adjusts the "boominess" of the room.

Parameter	Description
Decay Time	Sets the length of the Late Reverb. Range 1.5 to 30seconds
Early Reflections Level	Sets the volume of the initial wall reflections. Range: 0% to 100%
Early/Late Reverb Bal	Adjusts the balance between early refections and late reverb. Range: 0% to 100%
Late Reverb Delay	Sets the time between early reflections and the onset of the late reverb cloud. Range: 1ms to 350ms
Diffusion	Sets the amount of scattering of the late reverb cloud. Range: 0% to 100%
High Freq. Damping	Sets the rate at which high frequencies die away. Range: -10.0 to +3.0 damping factor
Low Freq. Damping	Sets the rate at which low frequencies die away. Range: -10.0 to +3.0 damping factor

Vocal Morpher

This unique effect allows you to select two vocal phonemes and morph between them using an LFO. Phonemes are the consonants and vowels we use in articulating speech sounds and these sounds are very distinctive and evocative. 30 different phonemes are available and these can be shifted up or down in pitch for even more effects.

To use the Vocal Morpher, you just select Phoneme A and Phoneme B from the list of thirty. Now the LFO automatically morphs back and forth between the two selected phonemes, creating interesting vocal articulations. The rate of the LFO is adjustable and you can select between Sine, Triangle or Sawtooth waveforms. The sine and triangle waves fade smoothly. The sawtooth wave gradually fades, then jumps abruptly back.

When the frequency of the A or B Phonemes is shifted up or down, entirely new effects can be produced. These frequency controls can also be used to tune the phoneme frequencies to the range of audio you are processing.
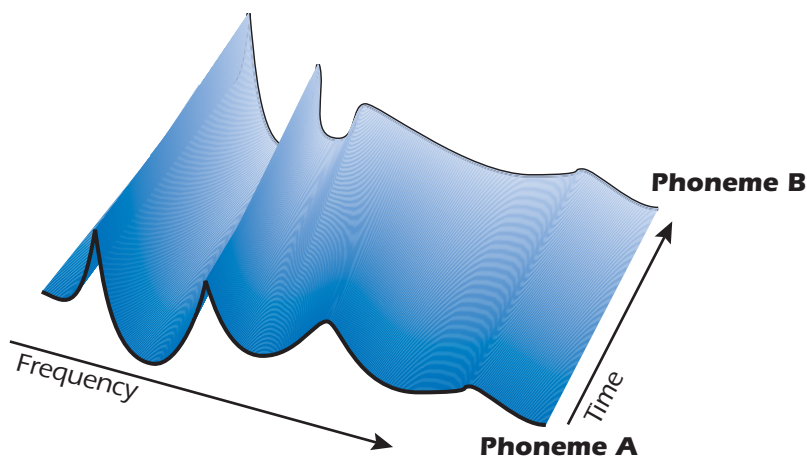
List of Available Phonemes

A	E	I	O	U	AA
AE	AH	AO	EH	ER	IH
IY	UH	UW	B	D	F
G	J	K	L	M	N
P	R	S	T	V	Z

Parameter	Description
Phoneme A	Select any of the available Phonemes for Phoneme A.
Phoneme A Tuning	Adjusts the frequency of Phoneme A up or down 2 octaves in semitone intervals. Range: -24 semitones to +24 semitones
Phoneme B	Select any of the available Phonemes for Phoneme B.
Phoneme B Tuning	Adjusts the frequency of Phoneme B up or down 2 octaves in semitone intervals. Range: -24 semitones to +24 semitones
LFO Rate	Controls how fast the phonemes morph back and forth. Range: .01Hz to 10Hz
LFO Waveform	Selects the waveform for the morph: Sinusoid, Triangle, Sawtooth

E-MU PowerFX

The hardware-accelerated effects of the E-MU Digital Audio System can also be used as VST inserts in Cubase LE. E-MU PowerFX allow you to use PatchMix DSP effects from within Cubase LE with minimal load on your CPU.

E-MU PowerFX incorporate smart time alignment technology which automatically compensates for system latencies and ensures proper synchronization of audio throughout the VST chain (if the host application supports this feature).

▼ E-MU PowerFX are not available at 96kHz and 192kHz sample rates.

❖ Cubase SX/SL/LE 2.0, Nuendo and Sonar (using the Cakewalk VST adapter 4.4.1) implement VST 2.X auto delay compensation.

E-MU PowerFX On/Off   Input Signal Present   FX Parameters   Preferences

FX Palette   FX Inserts   Output Signal Present   FX Presets   Preset Editing
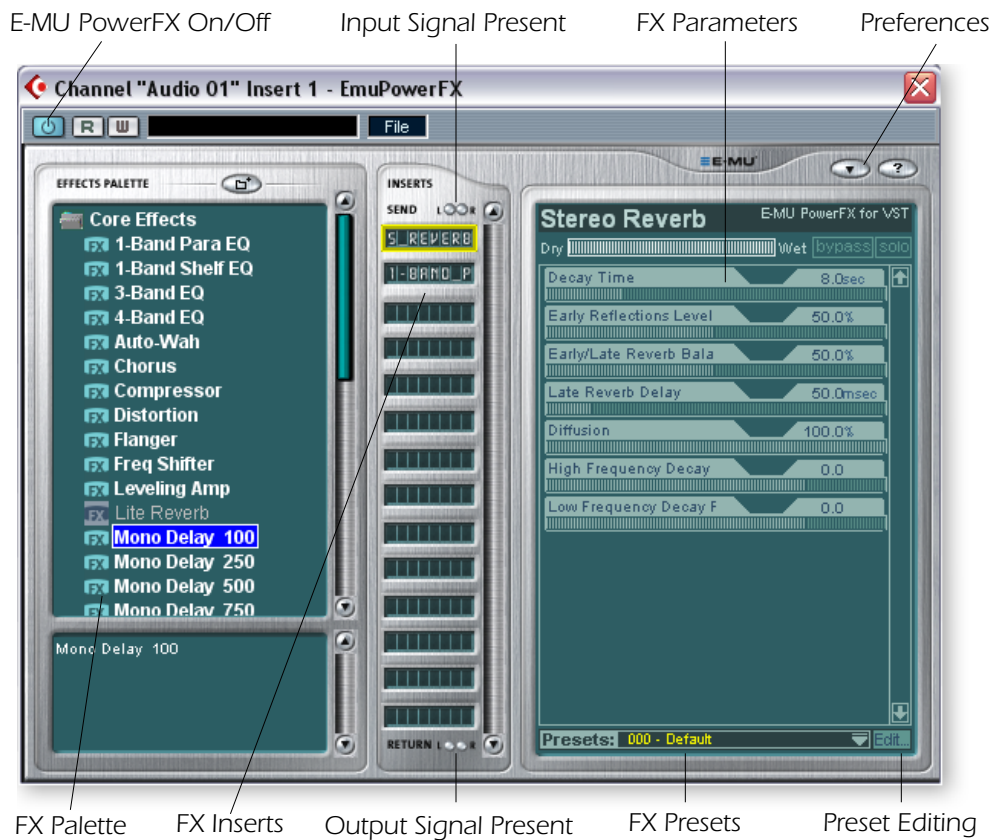
Parameter	Description
PowerFX On/Off	Enables or bypasses E-MU PowerFX.
FX Palette	Select from a single "Core" effect or a Multi -Effect.
FX Inserts	Drop Effects from the FX Palette here.
Signal Present LEDs	These indicators turn blue to show the presence of input and output signals.
FX Parameters	Select the desired effect in the center insert section, then adjust the wet/dry mix and parameters for the effect.
FX Presets	Select from the list of preprogrammed effect presets here.
Preset Editing	Click here to Save, Delete, Rename or Overwrite a User Preset. See the "User Preset Section" for more information

Parameter	Description
Preferences	*The Preferences menu allows you to:* ▪ *Toggle the Tooltips On or Off* ▪ *Extra Buffers - Check this box if excessive stuttering occurs when using E-MU PowerFX in your VST Host application. This box should be checked when using Fruity Loops.* ▪ *Render Mode - Induces realtime rendering in applications which do not support realtime rendering (WaveLab, Sound-Forge).*
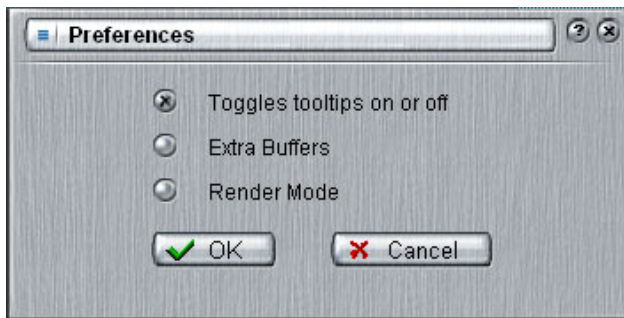
▶ **To Setup & Use E-MU PowerFX:**

Setup Cubase LE

1. Launch Cubase LE.

2. Instantiate **E-MU PowerF**X in an Insert or Aux Send location within Cubase (go to the EMU folder in VST plug-ins).

3. Press the Insert Edit button [e] in Cubase to bring up the E-MU PowerFX plug-in window shown on the previous page.

E-MU PowerFX

4. Make sure the **Insert Enable** button is illuminated, indicating E-MU Power FX is on. The blue "Signal Present" indicators will be illuminated if E-MU PowerFX is properly patched into a signal path.

5. Drag the desired effects from the Effects Palette to the center Insert strip.

6. Click on the Effect you wish to edit in the center Insert Strip *(it will be highlighted in yellow)*, then adjust the effects parameters in the right section of the window.

7. You can also select or edit User Presets from the section below the FX parameters. See the "User Preset Section" for more information.

Delay Compensation

If you are using Cubase VST 5.1, or another older sequencer without automatic delay compensataion, you will have to insert an E-Delay Compensator into any other audio tracks to keep them time-aligned.

8. Simply insert an E-Delay Compensator plug-in into the same insert location you used for E-MU PowerFX on any other audio tracks. That's it.

▼ *Using any driver other than "E-MU ASIO" may produce undesirable results when using E-MU PowerFX.*

Automating E-MU PowerFX

E-MU PowerFX can be automated in Cubase LE (or other recording host) just like any other VST effect. When "Write Automation" is activated in Cubase LE, control changes made in the PowerFX window during playback will be recorded on a special "Audio Mix" track located at the bottom of the Arrange Window. When "Automation Read" is activated, the recorded control changes will be played back.

▶ **To Record PowerFX parameter changes in Cubase LE**

1. Add E-MU PowerFX as a Channel Insert.

2. Rewind the song and enable "Automation Write" by pressing the **WRITE** button
   ▣   on, illuminating it. (Refers to Cubase LE. If you are using another application, refer to the documentation.)

3. Bring the E-MU PowerFX window to the front and select the Effect you want to automate. The effect parameters appear in the TV screen. Make sure the blue "On" button is lit.

4. Press the **Play** button on the Cubase LE Transport control. The song begins playing.

5. Adjust the E-MU PowerFX controls to achieve the effect you want. Rewind the song when finished.

6. Disable "Automation Write" and enable "Automation Read" ▣ . Playback the song to hear and view your changes.

7. To edit Automation, first enable both "Automation Write" and "Automation Read" and press Play. Cubase LE begins overwriting as soon as you change a control.

8. If you don't like the results and want to try again, select Show Used Automation from the Project menu. The Automation Subtrack appears. Next, click in the Parameter Display and select Remove Parameter.

   Note: This only erases one automation parameter from the Automation Subtrack. To erase multiple control edits, repeat the procedure above. See the Cubase LE manual for more specific information about automation editing.

▼ *Steinberg Cubasis does not have the control automation feature.*

▼ *Once you have recorded or drawn automation, do not delete or move effects from the Insert Strip. Doing so will result in unpredictable behavior.*

E-MU PowerFX Resource Availability

Because different collections of VST plug-ins and PatchMix Sessions can be run simultaneously, it is possible to load a Cubase Song or PatchMix Session for which resources are not available. If DSP resources are NOT available for an existing setup:

• E-MU PowerFX loads a Hardware I/O Path and simply passes audio through without any effects. The effects insert slot(s) in E-MU PowerFX will be "redded out".

• If no Hardware I/O Paths are available, the plug-in will be disabled and run in a soft pass-through mode. The effects insert slot(s) in E-MU PowerFX will be "grayed out".

• If DSP resources ARE available, but no Hardware I/O Paths are available, the plug-in will run in soft pass-through mode.

• If the sample rate is changed in the middle of a E-MU PowerFX session, E-MU PowerFX plug-ins will be bypassed, since the hardware effects cannot operate at 96kHz or 192kHz.

E-MU PowerFX Compatibility Chart

Application Name	Compatible?	Note	Render	Extra Buffers
Steinberg Cubase VST 5.1	Yes		Off	Off
Steinberg Cubase SX 1	Yes		Off	Off
Steinberg Cubase SX 2	Yes	Instrument Freeze triggers error if not in render mode.	Off	Off
Steinberg Cubase LE	Yes		Off	Off
Steinberg Cubase SL	Yes		Off	Off
Steinberg WaveLab 4	Yes		On	Off
Steinberg WaveLab Lite (ver 4)	Yes		On	Off
Steinberg WaveLab 5	No	Pops & clicks may occur. (Try 8 buffers at 1024)	On	Either
Sony Acid 4	Yes		On	Off
Sony Vegas 5	Yes		On	Off
Sony SoundForge 7	No	Power FX crashes when launched.	On	Off
Adobe Audition 1.5	No	Audio distortion & immediate lockup.	Any	Any
FruityLoops Studio 4.5	Yes		Off	On
Ableton Live 3.5	No	Distortion when FX parameters are changed.	On	Off
Cakewalk Sonar 3	Yes		Off	Off

Rendering Audio with E-MU PowerFX

Rendering (sometimes called Export) is a mixdown process performed by the host application, which creates a new digital audio file from a multitrack song. Rendering allows a virtually unlimited number of VST effects to be used because the audio processing is performed out of realtime.

E-MU PowerFX and the PatchMix DSP effects are strictly realtime processes. When E-MU PowerFX are used while rendering audio, the rendering process must proceed at realtime rate. Some host applications are not designed to handle realtime rendering and this can cause problems. E-MU PowerFX can be used with these applications if you are willing to follow certain guidelines.

General Tips for Rendering using PowerFX

- If an error message occurs, increase the "ASIO Buffer Latency" setting located in the device Setup dialog box. Depending on your setup, you may have to increase or decrease the Buffer Latency settings to find the setting that works.

- Instead of rendering with E-MU PowerFX, bounce the E-MU PowerFX processed tracks to another track in realtime.

- Check "Realtime Render" in the Render dialog box when using Cubase LE, Cubase SX2 or Cubase SL2. This setting will give the best results.

Tips for using Freeze Mode on Cubase LE

- Make the project length as short as possible. Freeze always renders the entire project length, even if the MIDI track being rendered is shorter.

- **Great Tip:** Temporarily bypass E-MU PowerFX (and any other effects) even when "Freezing" another track. This will allow the track to Freeze faster than realtime.

Using E-MU PowerFX with WaveLab and SoundForge

Stuttering in the audio can occur when rendering with SoundForge or any version of Steinberg WaveLab. This problem is caused by discontinuities in the first few audio buffers as they are fed by WaveLab to E-MU PowerFX. The problem can be eliminated by following these guidelines.

- Check "Render Mode" box in the E-MU PowerFX preferences.

- We recommend that you only use the MME/WAVE E-DSP Wave [xxxx] drivers.

- Reduce the "Buffer Size" in the WaveLab, Audio Preferences dialog box. This moves the stuttering to beginning of the file.

- Pad the beginning (and/or end) of your audio file with silence (.5 to several seconds depending on the file). This action causes the buffer discontinuities to occur *before* the song begins.

E-MU E-Wire VST

E-Wire is a special VST/ASIO Bridge which allows you to route digital audio via ASIO to PatchMix and back again.

E-Wire VST incorporates smart time alignment technology that automatically compensates for system latencies and ensures proper synchronization of audio throughout the VST chain. In addition, E-Wire also allows you to insert outboard audio gear into the VST environment.

E-Wire has three main components:

- A VST plug-in which handles the audio routing to PatchMix DSP.

- An ASIO mixer strip in PatchMix DSP configured to route audio to the E-Wire plug-in. You simply drop the effects you want to use into this strip.

- For hosts that don't support automatic delay compensation, a manual delay-compensation plug-in can be inserted in Cubase tracks or channels that don't use E-Wire to compensate for ASIO delay.

The diagram below may give you a better idea of how E-Wire works:

▪▪ **Note:** *It's easier to use E-MU PowerFX instead of E-Wire if you just want to use the hardware effects. (E-Wire was the precursor to E-MU PowerFX.) However, E-Wire can be very useful because it allows you to route VST inserts or Sends to Physical Inputs and Outputs via PatchMix DSP.*
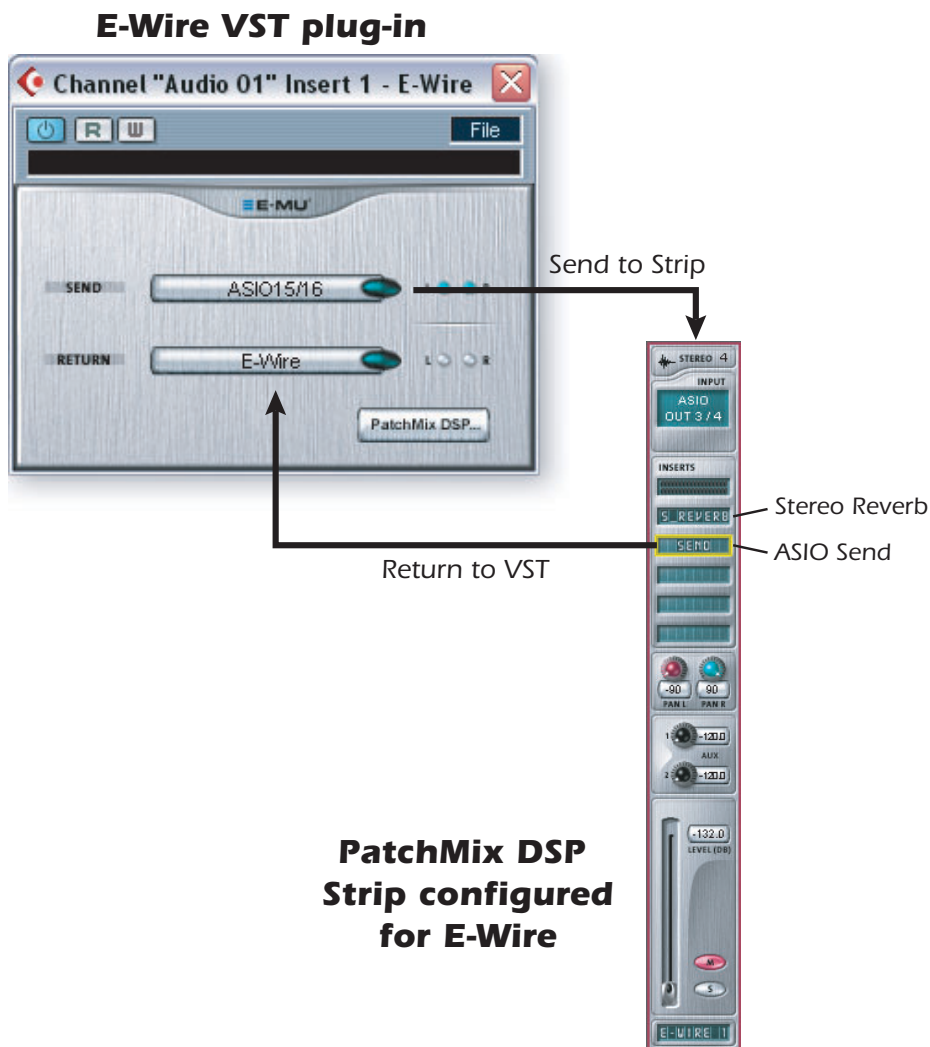
E-Wire bridges the gap between hardware I/O and the VST world. The E-Wire VST plug-in sends audio to a strip containing the desired effect. An ASIO Send routes the audio back to E-Wire VST.

▶ **To Setup and use E-Wire:**

Setup PatchMix DSP

1. Open PatchMix DSP application.

2. Insert an ASIO Input mixer strip into PatchMix DSP. (Alternately, you can select "New Session", select "E-Wire Example" and skip to step 6.)

3. Mute the strip or turn the Fader all the way down.

4. Insert an ASIO Send plug-in into one of the inserts on your ASIO strip.

5. Name your ASIO strip as an E-Wire strip.

6. Insert the desired PatchMix DSP effects into slots above the ASIO Send.

7. Save the Session.

Setup Cubase LE

8. Launch Cubase LE.

9. Instantiate E-Wire VST in an Insert or Aux Send location within Cubase.

10. Edit the E-Wire plug-in and activate the plug-in by pressing the blue button.

11. Set the ASIO Send and Return on the E-Wire plug-in to match the strip you set up for E-Wire.

12. Done.

E-Delay Compensation

An E-Delay Compensator must be inserted into any other audio tracks that are not using E-Wire in order to keep them time-aligned.

13. Simply insert an E-Delay Compensator plug-in into the same insert location you used for E-Wire on any other audio tracks. That's it.

E-Delay Compensator

As audio is transferred back and forth between the VST host application and the E-MU sound hardware, a delay in the audio stream is incurred. Normally this delay is compensated for automatically by the host application, but not all VST host applications support this automatic compensation.

A host will support PowerFX and E-Wire's plug-in delay compensation if it supports the SetInitialDelay feature of the VST 2.0 specification.

Currently automatic delay compensation is supported by the Steinberg 2.0 family (Nuendo 2.x, Cubase SX 2.0, Cubase LE 2.0,), Magix Samplitude 7.x, and Sonar (using the Cakewalk VST adapter 4.4.1), but not by Steinberg Cubase VST 5.1 and Cubasis.

The E-Delay Compensator utility plug-in is used to manually compensate for the transfer delay for hosts that DO NOT support plug-in delay compensation.

The E-Delay Compensator plug-in is used to delay the "dry" tracks (tracks without a PowerFX or E-Wire as an insert effect) or auxiliary (send) channels. For each dry track or send, add an E-Delay Compensator plug-in to re-align the track. The E-Delay Compensator is automatic and requires no user interaction to operate.

For example, consider a Cubase VST session with two audio tracks. If PowerFX or E-Wire is applied as an insert effect to the first audio track, but not to the second, the first track will be delayed in relation to the second track. The E-Delay Compensator should be added as an insert effect on the second track in order to provide delay compensation.
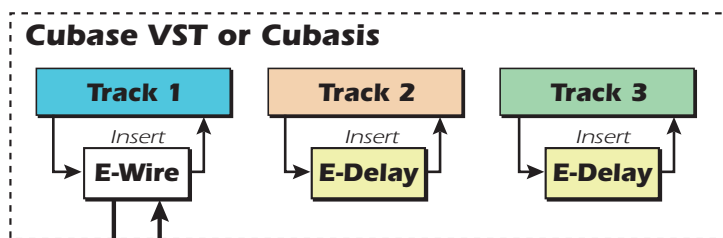
E-Delay Compensator Use

For host applications that don't support automatic delay compensation.

1. An E-Delay Compensator should be used when unprocessed audio tracks are played alongside tracks using a PowerFX or E-Wire plug-in.

2. Simply insert an E-Delay Compensator into each track that doesn't use a PowerFX or E-Wire send.

E-Delay Units Parameter

The Units value in the E-Delay dialog box should be set for the number of times you send ASIO down to the PatchMix DSP mixer and back in a single track. A single PowerFX insert chain with any number of effects only requires one delay unit because there was only one trip to the hardware and back. If you use two Cubase inserts in series on a track both using PowerFX or E-Wire, you would set the number parameter to 2 on all other audio tracks. Each trip down to PatchMix DSP and back to Cubase equals one unit.

In practical use, however, you'll probably never need to use more than one E-Wire VST on a single track since PowerFX effects can be placed in series. We have included this feature "just in case" you need it.

Here's one more example of how to use the E-Delay Compensator with different numbers of PowerFX/E-Wire sends on each track. The delay compensation on each track must equal the track with the maximum number of PowerFX/E-Wire sends. See the diagram below.
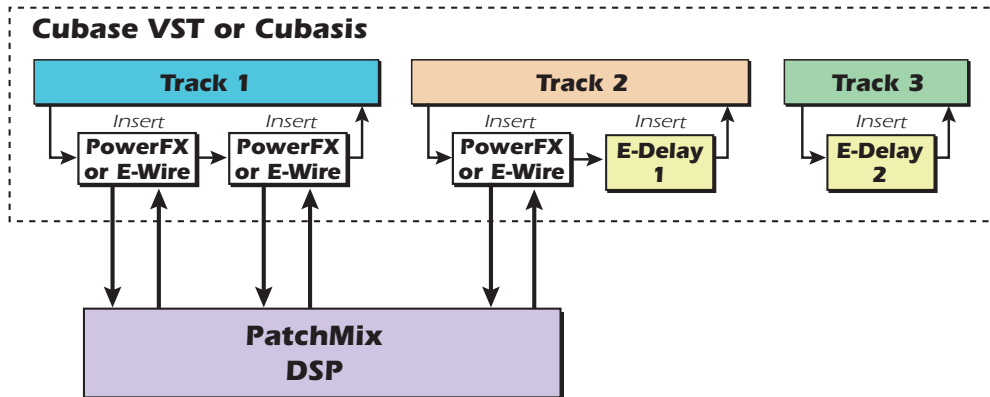
Since track 1 uses two PowerFX/E-Wire inserts, the delay of all the other tracks must equal two. Track 2 has one PowerFX/E-Wire insert and so adding one unit of E-Delay keeps it time aligned. Track 3 doesn't use a PowerFX/E-Wire insert and so it needs two E-Delay Units to remain in alignment.

Grouping Tracks

When several tracks require E-Delay Compensation, you can send the output of each track to a group or bus and use a single E-Delay Compensator on the output of the group or bus.

- E-MU Digital Audio System and PatchMix DSP must be installed.

- E-Wire is compatible with Cubase SX/SL/LE, Cubase VST, Wavelab, and Cakewalk Sonar (via DirectX-VST adapter) among others.

6 - Using High Sample Rates

Overview

When operating at 88.2k, 96k, 176.4k and 192k sample rates, the mixer functionality and number of I/O channels are reduced. These changes are summarized in the following tables. All S/PDIF inputs and outputs are disabled at 176.4kHz and192kHz. The number of ADAT channels also decreases at the 88.2k/96k and 176.4k/192k sample rates (due to the bandwidth limitations of the optical components).

When using 88.2kHz, 96kHz, 176.4kHz or 196kHz sample rates:

- **Effect processors are disabled.** *(Output sends & returns are still available.)*

- **ADAT is reduced to 4 chan. at 88k/96k, & 2 chan. at 176k/192k.**

- **ASIO channels are reduced to 8 stereo ASIO channels at 88k/96k, and 4 stereo ASIO channels at 176k/192k.**

- **At 176.4k/192k, the number of physical inputs/outputs is reduced.**

- **At the 176.4k & 192k sample rates, S/PDIF optical is disabled.**

The ADAT optical interface was originally designed to carry 8 channels at a 48kHz sample rate. We use the Sonorus® S/MUX™ standard to encode audio with higher sample rates onto the ADAT light pipe. In this multiplexing scheme, two ADAT channels are used to carry one 88.2k or 96k stream and four ADAT channels are used to carry one 176k or 192k audio stream. In order to use the ADAT interface at these higher sample rates, you must have other equipment that supports the Sonorus S/MUX standard.

E-MU 1820 System at 88.2k/96k (1010 PCI Card & AudioDock)

All outputs remain active at 88.2k/96k, but the number of ADAT channels is reduced from eight to four (see above). There are two possible input configurations when using the E-MU 1820 system at 88.2k/96k as shown in the chart below. Basically, you have the option of using the (4) ADAT input channels or (4) Line Inputs (Line Inputs 2 and 3). At 88.2kHz/96kHz the headphone output parallels the Monitor output and is no longer independently assignable.

E-MU 1820 Inputs/Outputs at 88.2kHz or 96kHz

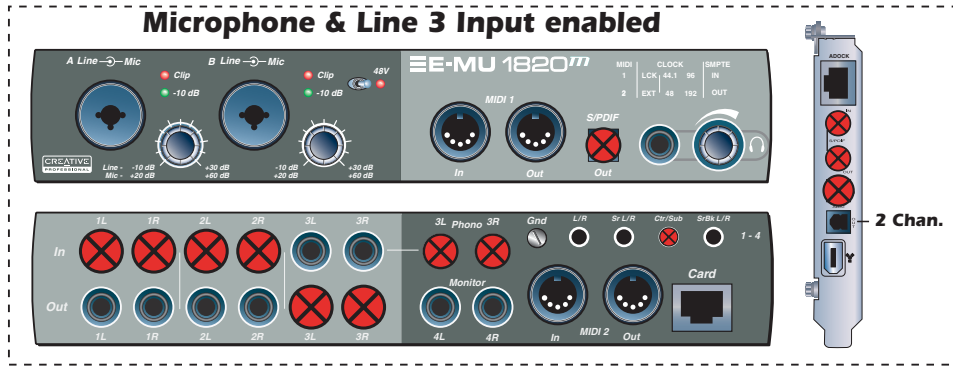| Source	Inputs (ADAT Option)	Inputs (Line In Option)	Outputs
ADAT	4	0	4
S/PDIF 1	2	2	2
S/PDIF 2	2	2	2
Microphone	2	2	-
Line 1	2	2	2
Line 2	0	2	2
Line 3	0	2	2
Line 4 out (monitor)	-	-	2
Headphone out	-	-	2 (monitor)
Total	12	12	18

▪ **Note:** *Headphone Out is permanently linked to Monitor Out in 96kHz mode.*

Input/Output - 88.2 kHz/96 kHz

ADAT enabled

or...

Line In 2 & 3 enabled
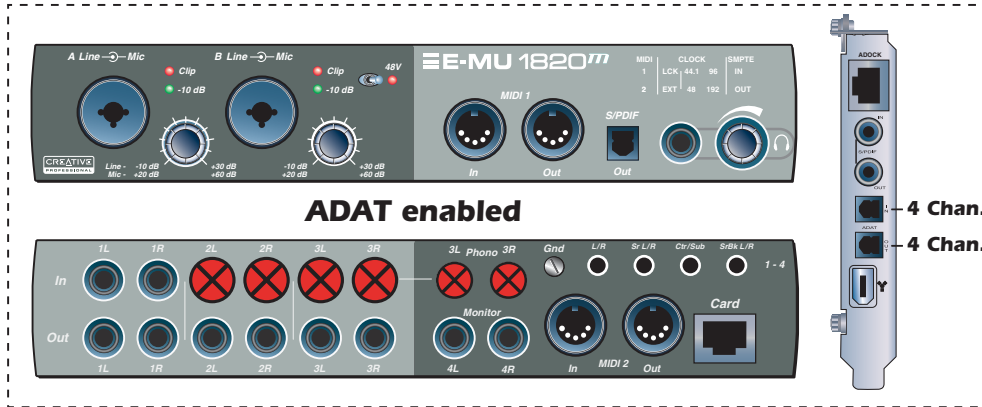
At the 88.2kHz/96kHz sample rates, all outputs are available but 4 inputs are lost. ADAT optical is also reduced to four channels. You have the option to enable either:

- *ADAT Inputs*
  or…
- *Line Inputs 2 and 3*

E-MU 1212M System at 88.2k or 96k (1010 PCI Card & I/O Card)

When using the E-MU 1212 system at 88.2kHz or 96kHz you have two analog inputs and outputs and two S/PDIF inputs and outputs. The ADAT input/output channels are reduced from eight to four using the S/MUX standard.

E-MU 1212M Inputs/Outputs at 88.2kHz or 96kHz

Source	Inputs	Outputs
ADAT	4	4
S/PDIF	2	2
Line	2	2
Total	8	8

E-MU 1820 System at 176.4kHz or 192kHz
(1010 PCI Card & AudioDock)

At the highest sample rate you have 4 inputs and 10 output channels.

There are four possible input configurations when using the E-MU 1820 system at 176.4kHz or 192kHz. Each of the three options provides four input channels.

- Microphone Input and Line 2 Input enabled
- Microphone Input and ADAT Input enabled *(2-chan ADAT)*
- Line 1 Input and ADAT Input enabled *(2-chan ADAT)*
- Line Inputs 1 & 3 enabled (allows use of turntable inputs at 192kHz)

S/PDIF is not specified to operate at 176.4kHz/192kHz and so all S/PDIF inputs and outputs are disabled. The headphone output parallels the Monitor output and is no longer independently assignable, just like at 96kHz.

E-MU 1820 Inputs/Outputs at 176.4kHz or 192kHz

Source	Inputs Mic & Line 3	Inputs Mic & ADAT	Inputs Line 1 & ADAT	Inputs Line 1 & 3	Total Outputs
ADAT	0	2	2	0	2
Microphone	2	2	0	0	-
Line 1	0	0	2	2	2
Line 2	0	0	0	0	2
Line 3	2	0	0	2	0
Line 4 out (monitor)	-	-	-	-	2
S/PDIF 1	0	0	0	0	0
S/PDIF 2	0	0	0	0	0
Headphone out	-	-	-	-	2 (monitor)
Total	4	4	4	4	10

E-MU 1212 System at 176.4k/192k (1010 PCI Card & I/O Card)

At 176.4kHz or 192kHz, you have two 24-bit inputs and outputs. S/PDIF is not specified to operate at these rates and so all S/PDIF inputs and outputs are disabled. The ADAT input/output channels are reduced to two channels (S/MUX standard).

E-MU 1212M Inputs/Outputs at 176.4kHz/192kHz

Source	Inputs	Outputs
ADAT	2	2
S/PDIF 1	0	0
Line	2	2
Total	4	4

Microphone & Line 3 Input enabled

2 Chan.

or...

Microphone & ADAT Input enabled

2 Chan.
2 Chan.

or...

Line Input 1 & ADAT Input enabled

2 Chan.
2 Chan.

or...

Line Inputs 1 & 3 enabled

2 Chan.
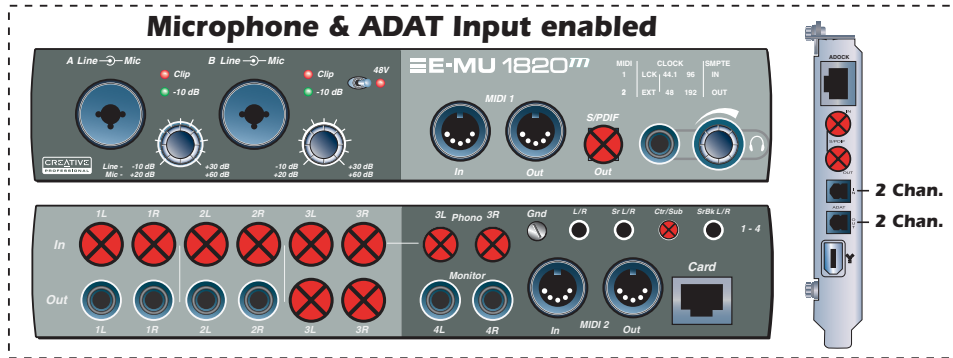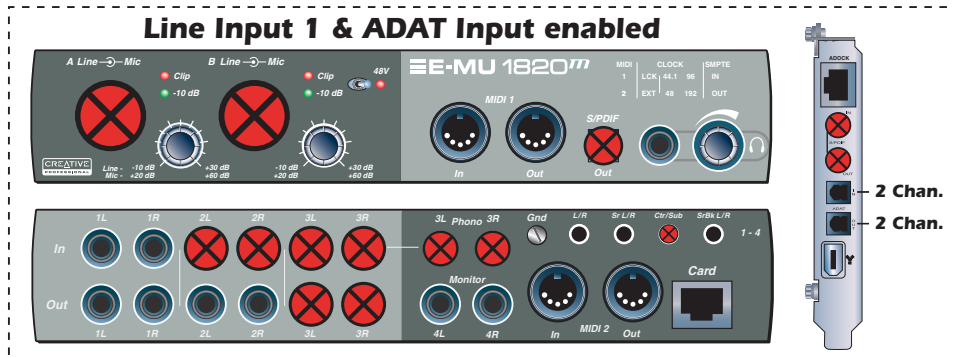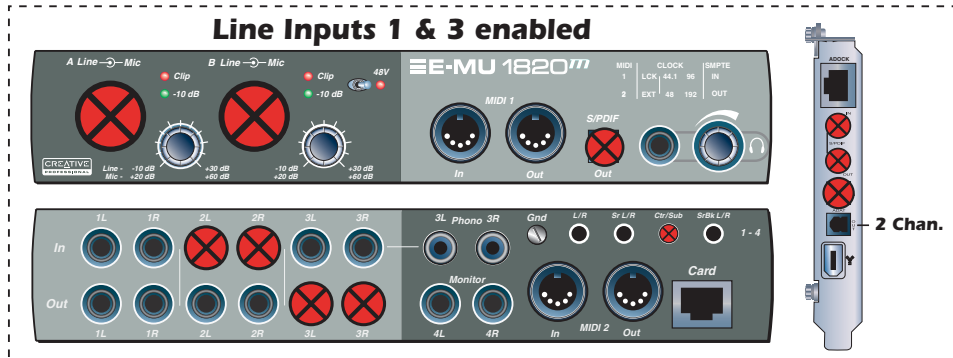
At the 176.4kHz or 192kHz sample rates, you sacrifice S/PDIF, line input 3, and line output 3. ADAT optical is reduced to two channels. You can choose one of the following options:

- *Microphone Inputs & Line 2 Inputs*
- *Microphone Inputs & ADAT Inputs*
- *Line Inputs 1 & ADAT*
- *Line Inputs 1 & 3*

WDM Recording and Playback Behavior

WDM recording and playback is supported at all PatchMix sample rates. The behavior of the driver with respect to PatchMix sample rate is described below.

When PatchMix and the WDM audio content (.WAV file format, playback and record settings in WaveLab. etc.) are both running at the same sample rate, and when a Wave strip or send is present in the PatchMix mixer configuration, WDM audio will be played or recorded "bit accurate" without sample rate conversion or bit truncation.

When running PatchMix at 44kHz/48kHz, if there is a mismatch between the WDM playback audio content and the PatchMix sample rate, sample rate conversion is performed, so that WDM audio will always be heard or recorded. Also, such non-native-sample-rate audio is truncated to 16-bits.

When running PatchMix at 88.2kHz, 96kHz, 176.4kHz or 192kHz, WDM record or playback audio content must be running at the same sample rate as PatchMix. If the sample rates are mismatched, NO AUDIO will be produced or recorded. In other words, the WDM driver does not perform sample rate conversion of any kind when PatchMix is running at 88.2kHz, 96kHz, 176.4kHz or 192kHz.

7 - Appendix

Sync Daughter Card Supplement

SMPTE Conversion

One of the main functions of the Sync Daughter Card is to convert SMPTE (LTC) to MIDI Time Code (MTC) and vice-versa. The term "Host MTC" refers to MTC, which is generated or used by the host application (Cubase, etc.). MTC is also available at the MIDI jack on the back of the Sync Card.

▼ **Warning:** *SMPTE and MTC do not provide sample sync for digital I/O. You must use Word Clock, S/PDIF or ADAT sync.*

SMPTE Features

- **Conversion of SMPTE to MTC quarter-frame messages & full-frame messages.**
  Constant quarter-frame messages are generated with steady SMPTE data input.
  Occasional MIDI full-frame messages are generated when SMPTE contains data dropouts.

- **Conversion of MTC (quarter frame & full frame) messages from the host computer to SMPTE out.**
  Simultaneous SMPTE and MTC output when receiving MTC from the host computer

- **Outputs SMPTE and MTC striping data.**
  SMPTE Start Time and type can be set via System Settings dialog box.

SMPTE Options

When the Sync Daughter Card is installed in your system a SMPTE button in the PatchMix DSP mixer becomes visible. Pressing the SMPTE button brings up the SMPTE window.
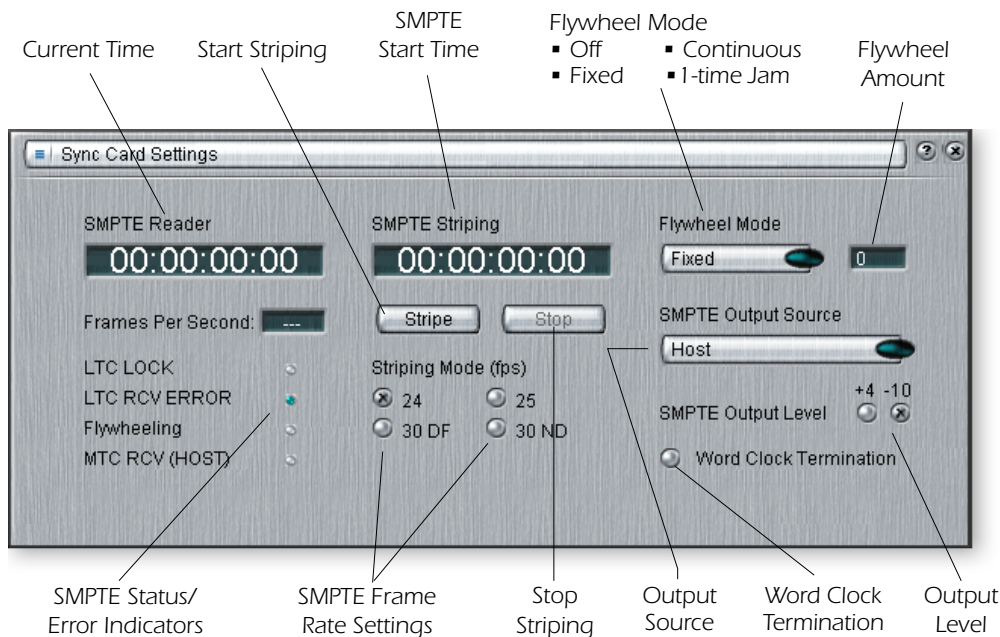
Mode (fps)	Sets the transmitted frame rate when striping SMPTE.
SMPTE Striping	Edit this field to set the start time in hours:minutes:seconds:frames for striping SMPTE.
Stripe Button	Initiates SMPTE Time Code generation at the SMPTE output beginning at the time set in the striping display.
Stop Button	Stops SMPTE striping. This button also stops SMPTE when One-Time Jam Sync has been initiated.
Flywheel Mode	Selects one of the four Flywheel modes. See the descriptions below.
Output Level	Sets the SMPTE output level from -10dBV (consumer) to +4dBu (pro).
FLY/JAM Frames	If flywheel mode is on and a dropout is detected this is the number of fly-wheel frames that will be output before the sync card stops and chases.
Word Clock Termination	Turns word clock termination on or off. Except in special cases, this control should normally be left on. See <u>"Word Clock In/Out"</u>.
SMPTE/MTC Output Source	This control selects the source of the SMPTE output jack. The choices are: Host MTC or the SMPTE Input jack (to regenerate SMPTE).

SMPTE Modes of Operation

Host Mode
The host computer is the source of synchronization. MTC messages are sent to the Sync Daughter Card from the computer application and converted into SMPTE. MTC is also output from the MIDI port on the Sync Daughter Card.

External Mode
SMPTE messages from SMPTE In are converted to MTC (quarter-frame messages) and sent to the host application. This happens automatically whenever LTC is received at the SMPTE input jack. Clean SMPTE data is also transmitted from SMPTE Out if "SMPTE (Regenerate)" is set.

Flywheel Mode
If the incoming SMPTE data is corrupted or missing frames, MTC code will continue to be output if "Flywheel mode" is enabled. The flywheel modes are described below.

Flywheel Modes

Off	Upon any dropout, MTC stops and the Sync card monitors the input for valid code. If valid code is again received, it chases and relocks.
Fixed 0-127	Upon any dropout, MTC continues outputting Quarter-frame messages at the same rate (flywheeling). When a dropout is detected, this is the number of frames that will be output before the Sync card stops output-ting MTC and monitors the input for valid code. If valid code is again received, it chases and relocks.
Continuous	Upon any dropout, MTC continues outputting Quarter-frame messages at the same rate (flywheeling). The Sync card monitors the input for valid code and continues flywheeling until valid code is received, then relocks.
1-Time Jam Sync	Upon any dropout, MTC continues outputting Quarter-frame messages at the same rate (flywheeling) without monitoring the SMPTE input until the Stop button is pressed.

Stripe Mode

This mode is used to record SMPTE time code onto an audio track of another recorder. SMPTE is output when the Start button is pressed in the System Settings menu and begins at the time set by the Start Time setting. MTC is also simultaneously output from the Sync Daughter Card MIDI out. SMPTE and MTC will continue to be output until the Stop button is pressed. See Striping SMPTE.

SMPTE Background

SMPTE time code was standardized way back in 1969 by the Society of Motion Picture and Television Engineers as a way to mark frame numbers on video tape.

Using SMPTE, a particular location can be precisely located by simply entering the appropriate time code number which is expressed in Hours, Minutes, Seconds, Frames and Subframes. This is possible because each frame of SMPTE time code contains absolute location information expressed in digital form.

There are two types of SMPTE time code: Vertical Interval Time Code (VITC), which is used on video tape and Longitudinal Time Code (LTC) or audio time code. VITC is strictly used for video and has the advantage of being able to be read while the video deck is paused. LTC can be recorded on the audio or sync tracks of video tape and can thus be used in audio or video work.

Longitudinal time code is the type of SMPTE used on the Sync Daughter Card. It contains 80 bits of information per frame. An audio SMPTE frame is divided into 80 "bit cells". A voltage change during a bit cell period constitutes a digital "1" and no change during a bit cell period constitutes a digital "0". In addition to the location bits, there are user bits that may contain information about tape reel numbers, bits dealing with video information, and a 16-bit sync word at the end of the frame.
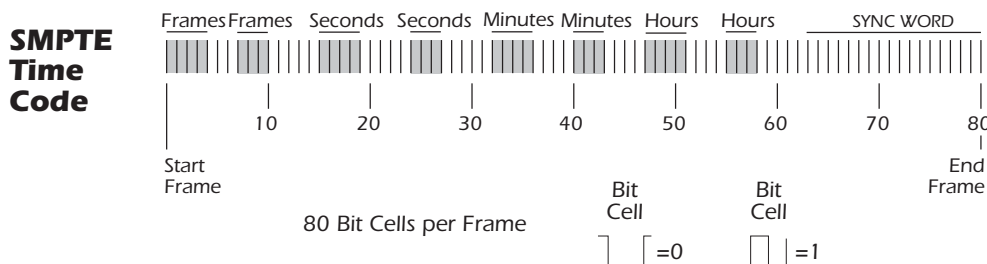
There are four types of SMPTE time code in general use: 24, 25, 30 frame-per-second and 30 drop-frame. In general, you should choose one rate (30 non-drop is common in audio) and stick with it for initial recording and later editing.

Types of SMPTE

Type	Use	Hours	Minutes	Seconds	Frames
24 frame	US Film	00-23	00-59	00-59	00-23
25 frame	Euro. Film + Video	00-23	00-59	00-59	00-24
30 drop-frame	US & Japan Color Video	00-23	00-59	00-59	00-29
30 non-drop	US & Japan B/W Video	00-23	00-59	00-59	00-29

The four frame rates are all straightforward except 30 drop-frame. The 30 df rate came about because the US color video frame rate is actually 29.97 frames/sec instead of 30 frames/sec. This adds up to an error of 108 frames each hour relative to "wall clock"

time! (A one hour program would actually be 59 minutes and 56.4 seconds long.) Drop frame was designed to correct this time difference. In 30 Drop Frame, every minute except **00-10-20-30-40-50** have the first two frames, 00 and 01, "dropped", hence the name drop-frame.

Why use SMPTE?

SMPTE sync, although well over 30 years old, has the advantage of being able to be recorded as an audio track. This allows it to be used with virtually any kind of recording equipment from tape recorders to computer-based digital audio recorders. You can even buy phonograph records with a SMPTE stripe!

SMPTE was designed in the days when tape dropout was a common occurrence and so it was designed to convey "absolute" location information. Since each frame of SMPTE code provides its own unique identification, it provides the ability for a receiving device to recover from data dropout. In addition, edits can be performed in the middle of a song with just a few seconds of pre-roll before the punch-in point. SMPTE is also standardized, which means that code generated on different makes of equipment will be compatible with each other. SMPTE also has fairly good resolution, especially at the subframe level. You'll be happy to know that the Sync Daughter Card resolves to the subframe level. The chart below shows subframe accuracy at the three frame rates.

SMPTE Subframe Resolution

Frames-per-second	Resolution
24 fps	.521 mS
25 fps	.500 mS
30 fps	.417 mS

Striping SMPTE

Printing SMPTE to a track is called striping (as in stripe). SMPTE time code is recorded on an unused audio track of another recorder, then played back into the Sync Daughter Card. The Sync Daughter Card passes the location information on to the host computer as MTC quarter-frame data to be used by an application such as an audio recorder or sequencer.

SMPTE is usually recorded at about -3 VU on semi-pro gear, -10 VU on professional gear and 0 VU on video gear. Experiment to find the optimum levels. When printing to a time code track of a video deck, be careful. The time code playback head locations on video decks are not standardized and can cause gross timing errors. Time code which is striped on an audio track will always be in sync with the picture. SMPTE code is traditionally recorded on the right channel of a video recorder.

Avoiding SMPTE problems

Problems in reading SMPTE time code can often be related to poor quality code on the tape. Poor quality code can be caused by a number of problems, the most common being dirty or misaligned heads, amplifier clipping, or too many generations of audio dubbing. Other problems can be caused by running the SMPTE signal through signal processing devices such as Limiters, Reverbs, Harmonizers, etc. (Don't laugh, it *has* been done!) In fact, many video decks have built in AGCs (Automatic Gain Controls) which will ruin the SMPTE signal if the input level is too high. Always check playback to insure that the time code is usable. In general, no signal processing should be used on the SMPTE signal. SMPTE code is delicate and should be treated as such.

Duplicating SMPTE time code

The Sync Daughter Card always generates clean SMPTE from the SMPTE output when reading SMPTE in. This time code is in sync with the incoming SMPTE and can be used to feed other devices in your studio or to clean up old SMPTE tracks. Copying SMPTE code from track to track produces deterioration of the signal with each generation, although one generation of dubbing will probably be OK.

Other Tips for using SMPTE

1. **Use ascending time code.** Jumps in the code are OK as long as the SMPTE code jumps forward in time as the tape moves forward in time. A good way to avoid any problems with this is to simply stripe the entire project with SMPTE before you record any other tracks.

2. **Allow enough leader.** Leave a few seconds between each song to allow SMPTE to sync up before the song starts.

Keep written logs. Keeping written records of song start points and edit cues can save time and avoid wasteful searching through a project that was recorded earlier.

Example SMPTE Connection

In the diagram below, Cubase is controlling the entire system by sending MTC to the Sync Card which converts MTC to SMPTE. SMPTE is fed to the ADAT/BRC to convey the absolute time information (hours-minutes-seconds-frames). ADAT/BRC is the word clock master, controlling the Digital Audio System either through the embedded clock in the ADAT optical stream or using word clock.
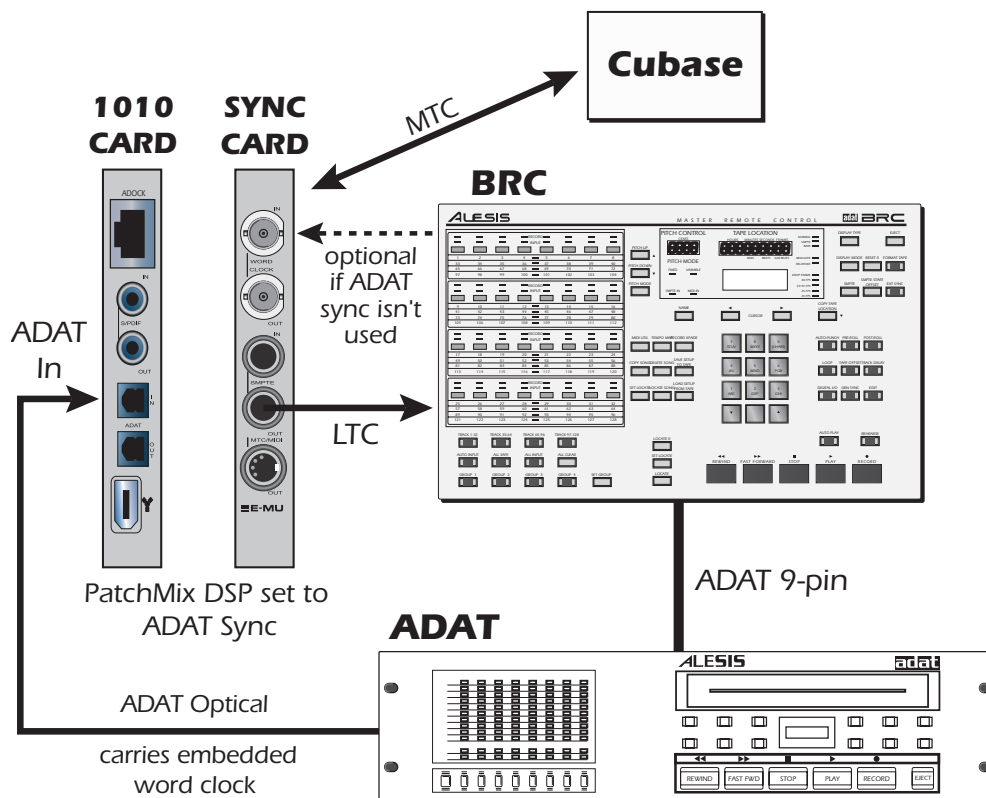
The Sync Card should not be used as both the SMPTE and word clock master. Word Clock is generated by the Digital Audio System and NOT by the software application (Cubase). SMPTE is not locked to Word Clock inside the Sync Card—they are completely independent.

MIDI Time Code (MTC)

MIDI time code is basically SMPTE time code adapted to the world of MIDI. MTC specifies "absolute" location information in hours:minutes:seconds:frames, just like SMPTE. There are two main kinds of messages in MTC: Full-frame messages and Quarter-frame messages.

Full-frame messages are ten bytes long and are sent when SMPTE start, stops, or relocates. Full-frame messages contain the entire SMPTE number of, hours, minutes, seconds, frames, as well as the SMPTE type: 24fps, 25fps, 30fps non-drop, 30fps drop.

Quarter-frame messages are sent at each quarter of a SMPTE frame and only carry 1/8th of the SMPTE time message. Quarter-frame messages require two entire SMPTE frames to send the complete time stamp (h:m:s:f). Timing accuracy is maintained as long as the quarter-frame messages continue to come in at a constant rate.

▶ **To Enable MTC:**

MIDI Time code disables the use of MIDI port 2 on the back panel of the AudioDock.

1. Open **Session Settings** from the toolbar.
2. Select the **MIDI** tab and choose **Sync Card/MTC** from the MIDI options.
3. Click **OK** to close the window.

Since it is important to have a stable timing reference for your song or sequence, we have given MTC its own MIDI output port on the Sync Daughter Card. This ensures that the timing information will not be affected by other MIDI data on the line.

▼ *MTC and SMPTE do NOT synchronize at the sample rate and are not locked to word clock in any way.*

SMPTE and MTC are used to synchronize music but do not have the required resolution to sample-lock digital audio.

Word Clock In/Out

Word clock provides a standardized means of synchronizing multiple digital audio devices so that data can be transferred digitally. **In order to digitally transfer from one device to another, the two devices MUST be synchronized.** Clicks and pops in the audio will result when transferring digital audio which is not synchronized.

The E-MU 1010 PCI card can be externally clocked from the ADAT input, S/PDIF input or from the Sync Daughter card (if installed). **In a digital studio, all digital devices in the system should run off the same master Word Clock**.

▶ **To Synchronize PatchMix DSP to an External Clock Source:**

1. Make sure an external clock source is connected to the E-MU Digital Audio System hardware via the word clock, ADAT or S/PDIF input.
2. Open the **Session Settings** dialog box.
3. Under the **System** tab, select **External Source**, then select either **word clock, ADAT** or **S/PDIF**.
4. Press **OK** to close the dialog box.
5. Check the **Sync** section of PatchMix DSP to verify that the **Locked** indicator is illuminated.

Devices can be connected in daisy chain fashion (word clock out connected to the next unit's word clock in) or in parallel for one or two devices, but professional digital studios normally use a master word clock generator or "House Sync" with a distribution system so that each device receives a phase-coherent and jitter-free word clock.

❖ *Word clock, ADAT and S/PDIF synchronize at the sample rate and are used to transfer digital data between machines.*
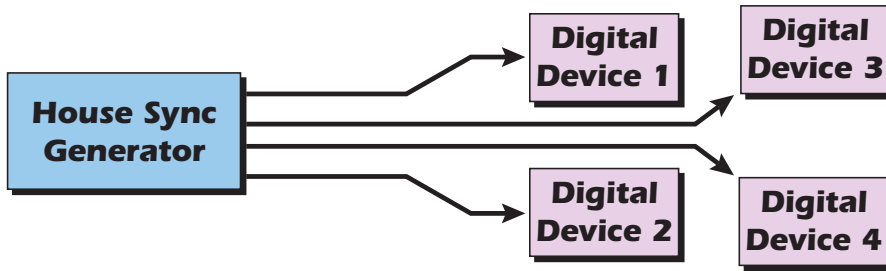
A master word clock generator is preferable for larger digital setups.

Word Clock In: Receives word clock (sample clock) from another digital device such as a digital video deck, digital recorder or digital mixer.

Word Clock Out: Sends word clock (sample clock) to another digital recorder. Word clock is always output, whether it is generated by the internal clock or passed through from the word clock input.

75Ω On/Off: Termination for the word clock input can be switched on or off in the Sync Card menu of the PatchMix DSP application. Normally word clock termination should be left on. If you have problems with a weak word clock signal, try turning termination off. See Word Clock Termination.

The diagram below shows the proper way to connect and terminate a serial word clock chain. Using a BNC "T" connector ensures that word clock is precisely in phase for both devices. The middle device has termination turned **Off** and the last device in the word clock chain has termination turned **On**.
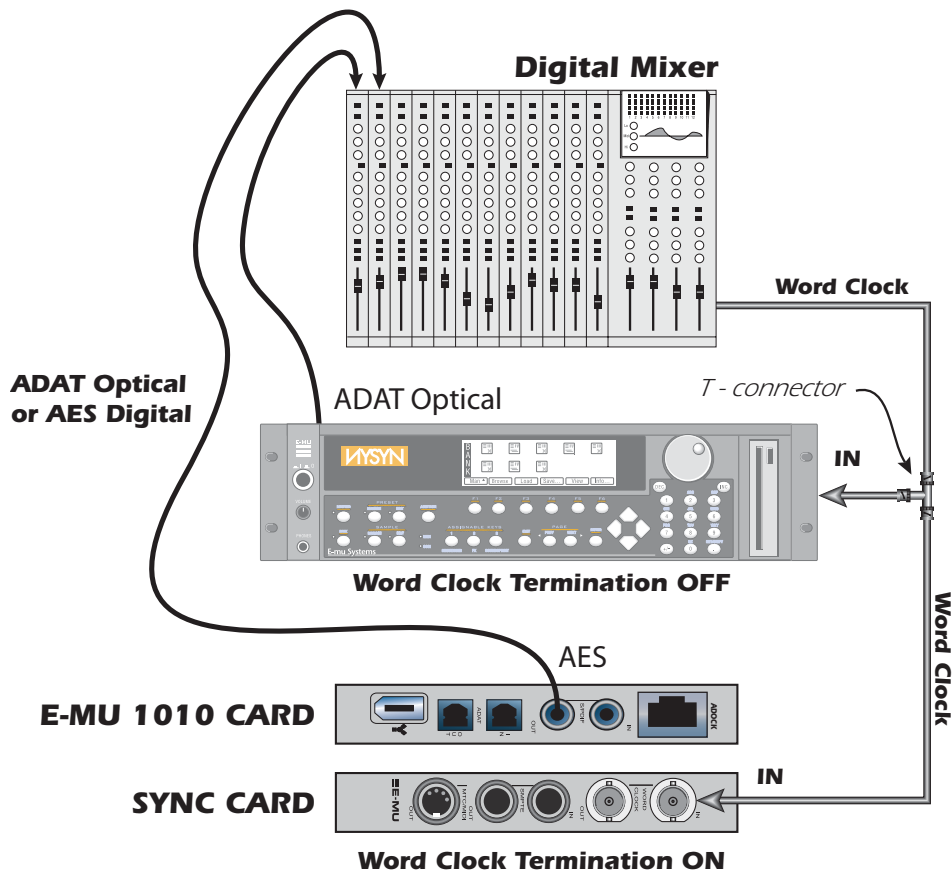
This diagram shows the proper way to connect word clock if you don't have a multi-output word clock generator. The last device in a Word Clock chain should have Termination ON.

Getting in Sync

Whenever you connect external digital audio devices together, you need to be aware of how they are synchronized to each other. Simply connecting digital out to digital in doesn't guarantee that two digital devices are synced, even if audio is being passed. Unless you have set one to be the Master and the other a Slave, they are probably NOT synchronized and the quality of your audio will suffer.

S/PDIF and ADAT are two commonly used digital audio formats. Both these digital formats carry an embedded word clock which can be used to synchronize the digital equipment. You must enable "External Clock" on the slave device to have clock sync!

The diagrams below show two ways to synchronize an external A/D - D/A converter to the E-MU Digital Audio System using the ADAT lightpipe connection.

In the first example, only the A/D converters on the external device are being used. Only one lightpipe is needed as long as PatchMix is set to receive its word clock signal from the external device. The external A/D is the Master and the E-MU DAS is the Slave.

External Device supplies Master Clock
(via ADAT)

The lightpipe carries eight channels of audio data and an embedded clock.

ADAT Out

Master

External A/D - D/A Converter

Set PatchMix DSP to receive:
External ADAT Sync

Slave

PatchMix DSP supplies Master Clock
(via ADAT)

This lightpipe carries eight channels of audio data.

Set External Device to receive:
External ADAT Sync

ADAT Out

Slave

ADAT In

External A/D - D/A Converter

This lightpipe carries an embedded clock signal & eight channels of audio.

Master

In the second example a second lightpipe is used to supply "embedded word clock", as well as eight channels of audio to the external A/D - D/A. The external device MUST be set to receive external clock via ADAT or the units will not be synchronized. The E-MU Digital Audio System is the Master and the external A/D - D/A is the Slave.

Useful Information

AES/EBU to S/PDIF Cable Adapter

This simple adapter cable allows you to receive AES/EBU digital audio via the S/PDIF input on the E-MU 1010 PCI card. This cable may also work to connect S/PDIF out from the 1010 PCI card to the AES/EBU input of other digital equipment.
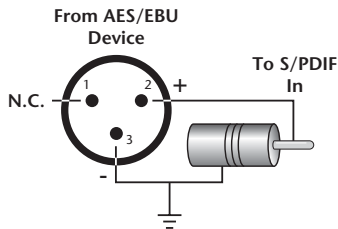
Cables - balanced or unbalanced?

All inputs and outputs on the E-MU Digital Audio System are designed to use either balanced or unbalanced cables. Balanced signals provide an additional +6dB of gain on the inputs and are recommended for best audio performance, although unbalanced cables are fine for most applications. If you're having problems with hum and noise or just want the best possible performance, use balanced cables.

Balanced Cables

Balanced cables are used in professional studios because they cancel out noise and interference. Connector plugs used on balanced cables are XLR (3-prong mic connector) or TRS (Tip, Ring, Sleeve) 1/4" phone plugs.

Balanced XLR Connectors
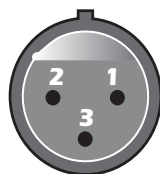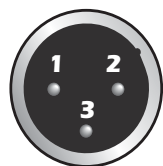
Input   **Output**

1 = Ground/shield
2 = Hot (+)
3 = Cold (-)

Balanced 1/4" TRS Connectors

Sleeve = Ground
Tip = Hot (+)
Ring = Cold (-)
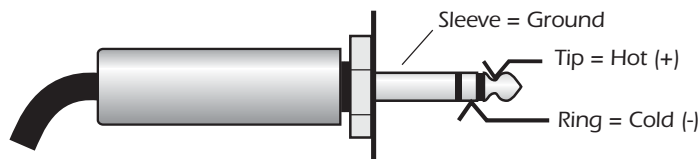
Unbalanced 1/4" Connectors

Sleeve = Ground
Tip = Signal

▼ **WARNING:** Do NOT use balanced audio cables when connecting balanced outputs to unbalanced inputs. Doing so can increase noise level and introduce hum. Use balanced (3-conductor) cables ONLY if you are connecting balanced inputs to balanced outputs.
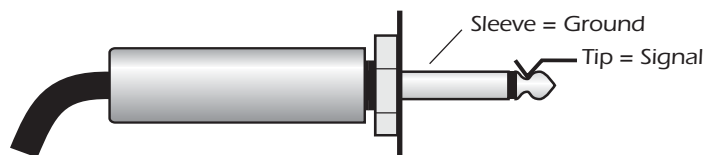
Balanced cables have one ground (shield) connection and two signal-carrying conductors of equal potential but opposite polarity. There is one "hot" or positive lead, and a "cold" or negative lead. At any point in time, both conductors are equal in voltage but opposite in polarity. Both leads may pick up interference, but because it is present both in and out of phase, this interference cancels out at the balanced input connection.

Unbalanced Cables

Unbalanced cables have one conductor and one ground (shield) and usually connect via unbalanced 1/4" phone plugs or RCA phono plugs. The shield stays at a constant ground potential while the signal in the center conductor varies in positive and negative voltage. The shield completely surrounds the center "hot" conductor and is connected to ground in order to intercept most of the electrical interference encountered by the cable. Unbalanced cables are more prone to hum and interference than balanced cables, but the shorter the cable, the less hum introduced into the system.

Digital Cables

Don't cheap out! Use high quality optical fiber (for ADAT) and low-capacitance electrical cables (for S/PDIF) when transferring digital I/O to avoid data corruption. It's also a good idea to keep digital cabling as short as possible (1.5 meters for plastic light pipes; 5 meters for high quality glass fiber light pipes).

Grounding

In order to obtain best results and lowest noise levels, make sure that your computer and any external audio devices are grounded to the same reference. This usually means that you should be using grounded AC cables on both systems and make sure that both systems are connected to the same grounded outlet. Failure to observe this common practice can result in a ground loop. 60 cycle hum in the audio signal is almost always caused by a ground loop.

Phantom Power

Phantom power is a dc voltage (+48 volts) which is normally used to power the preamplifier of a condenser microphone. Some direct boxes also use phantom power.

Pins 2 and 3 of the AudioDock microphone inputs each carry +48 volts dc referenced to pin 1. Pins 2 and 3 also carry the audio signal which "rides" on top of the constant 48 volts DC. Coupling capacitors at the input of the AudioDock block the +48 volt DC component before the signal is converted into digital form. The audio mutes for a second when phantom power is turned on. After turning phantom power off, wait two full minutes before recording to allow the DC bias to drain from the coupling capacitors or this bias could affect the audio headroom.
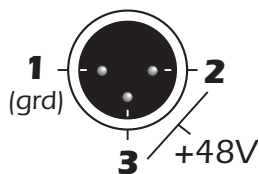
Balanced dynamic microphones are not affected by phantom power. An unbalanced dynamic microphone may not work properly, but will probably not be damaged if phantom power is left on.

Ribbon microphones should NOT be used with phantom power. Doing so can seriously damage the ribbon element.

Since ribbon microphones are fairly specialized and generally expensive, you'll know if you own one. Most microphones are either of dynamic or condenser type and these are not harmed by phantom power.

Appearance Settings in Windows

Adjusting the "Performance Options" in Windows will improve the screen appearance when moving the mixer around on the screen.

▶ **To Improve the Appearance Settings:**

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

Technical Specifications

Specifications: 1820M System

GENERAL

Sample Rates	44.1 kHz. 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz from internal crystal. Externally supplied clock from S/PDIF, ADAT (or word clock with optional Sync Card)
Bit Depth	16 or 24-bits
Hardware DSP	100MIPs custom audio DSP. PCI Bus-Mastering DMA subsystem reduces CPU usage. Zero-latency direct hardware monitoring with effects. 1394 Firewire Core - Texas Instruments
Converters & OpAmps	ADC - AK5394 (AKM) DAC - CS4398 (Cirrus Logic) OpAmp - NJM2068M (JRC)
WDM Drivers	8 channels — operational at 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz & 192kHz
AudioDockM Power Use	1.25A @ +12V 15W.

ANALOG LINE INPUTS

Type	Servo-balanced, DC-coupled, low-noise input circuitry
Level (software selectable)	Professional: +4 dBu nominal, 20 dBu maximum (balanced)
	Consumer: -10 dBV nominal, 6 dBV maximum (unbalanced)
Frequency Response	+/- .05dB, 20 Hz - 20 kHz
THD + N	-110 dB (.0003%) 1kHz at -1 dBFS
SNR	120 dB (A-weighted)
Dynamic Range	120 dB (A-weighted)
Channel Crosstalk	< -115 dB, (1 kHz signal at -1 dBFS)
Common-mode Rejection	> 40 dB at 60Hz
Input Impedance	10K ohm

ANALOG LINE OUTPUTS

Type	Balanced, low-noise, 2-pole low-pass differential filter
Level (software selectable)	Professional: +4dBu nominal, 20dBu (balanced)
	Consumer: -10dBV nominal, 6dBV maximum (unbalanced)
Frequency Response	+0.0/-0.35 dB, 20 Hz - 20 kHz
THD + N	-105 dB (.0006%) 1kHz signal at -1dBFS
SNR	120 dB (A-weighted)
Dynamic Range	120 dB (A-weighted)
Stereo Crosstalk	< -120 dB, 1kHz
Output Impedance	560 ohms

Specifications: 1820M System

MIC PREAMP/LINE INPUT

Type	TFPro™ combination microphone preamp and line input
Frequency Response	+0.8/-0.1 dB, 20 Hz - 20kHz
Stereo Crosstalk	< 120 dB, 1kHz

LINE INPUT

Gain Range:	-12 to +28 dB
Max Level:	-17 dbV (19.2 dBu)
THD+N:	-100 dB (.001%), 1 kHz at -1 dBFS
Dynamic Range:	107 dB (A-weighted, min. gain)
SNR:	107 dB (A-weighted, min. gain)
Input Impedance:	10K ohm
CMRR:	> 40 dB (60Hz)

MICROPHONE PREAMP

Gain Range:	-10 to +50 dB
Max Level:	-12 dbV (-9.8 dBu)
THD+N:	-100 dB (.001%), 1 kHz at -1 dBFS
SNR:	106 dB (A-weighted, min. gain)
Input Impedance:	330 ohm
CMRR:	> 80 dB (60Hz)

HEADPHONES

Frequency Response:	+0.0/-0.35 dB, 20 Hz - 20 kHz
THD+N: (1 kHz, max. level)	33 ohm load: -69 dB (0.035%) 600 ohm load: -94 dB (0.002%)
SNR:	117 dB (A-weighted)
Dynamic Range:	117 dB (A-weighted)
Stereo Crosstalk:	< -100 dB (1kHz at -1 dBFS, 600 ohm load)
Max Output Power:	500 mW
Output Impedance:	22 ohms
Gain Range:	85 dB

Specifications: 1820M System

TURNTABLE INPUT	RIAA equalized phono input
Frequency Response:	+/-0.5 dB, 50 Hz - 20 kHz
THD+N:	-76 dB (.015%) (1 kHz, 10 mV RMS unbalanced input)
SNR:	90 dB (1kHz, 10 mV RMS unbalanced input)
Stereo Crosstalk:	< -80 dB (1kHz at -1 dBFS)
Maximum Level:	Professional: 80 mV RMS Consumer: 20 mV RMS
Input Capacitance:	220 pF
Input Impedance:	47K ohm

DIGITAL I/O

S/PDIF	▪ 2 in/2 out coaxial (transformer coupled) ▪ 2 in/3 out optical (software switchable with ADAT) ▪ AES/EBU or S/PDIF (switchable under software control)
ADAT	▪ 8 channels, 24-bit @ 44.1/48 kHz ▪ 4 channels, 24-bit @ 96 kHz ▪ 2 channels, 24-bit @ 192 kHz
Firewire	400 IEEE 1394a port (6-pin) Compatible with DV cameras or HDs
MIDI	2 MIDI in, 2 MIDI out

SYNCHRONIZATION

Internal Crystal Sync:	44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz
	ADAT, S/PDIF (optical or coaxial)
	Word Clock (sync card only) - (75 ohm termination, switchable)
RMS JITTER @ 44.1K (Measured via Audio Precision 2)	SRSync SourceRMS jitter in picoseconds 44.1 kHz internal Crystal 596ps 44.1 kHz Optical Input 795ps
SMPTE	Converts to/from longitudinal time code (LTC) to MIDI time code (MTC)
Frame Rates	24, 25, 30 drop, 30 non-drop frames/second. Compatible with 29.97 fps timecode
Modes	Regeneration, stripe and conversion modes
Input Level:	0.5 - 4V p-p
Output Level:	+4 dBu, -10 dBV (software selectable)
Input Impedance:	10K ohm

Specifications: 1820 System

GENERAL

Sample Rates	44.1 kHz. 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz from internal crystal. Externally supplied clock from S/PDIF, ADAT (or word clock with optional Sync Card)
Bit Depth	16 or 24-bits
Hardware DSP	100MIPs custom audio DSP. PCI Bus-Mastering DMA subsystem reduces CPU usage. Zero-latency direct hardware monitoring with effects 1394 Firewire Core - Texas Instruments
Converters & OpAmps	ADC - PCM1804 (TI/Burr-Brown) DAC - CS4392 (Cirrus Logic) OpAmp - NJM2068M (JRC)
AudioDock Power Use	1.1A @ +12V 13W.

ANALOG LINE INPUTS

Type	Servo-balanced, DC-coupled, low-noise input circuitry
Level (software selectable)	Professional: +4 dBu nominal, 20 dBu maximum (balanced)
	Consumer: -10 dBV nominal, 6 dBV maximum (unbalanced)
Frequency Response	+0.0/-0.2 dB, 20 Hz - 20 kHz
THD + N	-102 dB (.0008%) 1kHz at -1 dBFS
SNR	111 dB (A-weighted)
Dynamic Range	111 dB (A-weighted)
Channel Crosstalk	< -115 dB, (1 kHz signal at -1 dBFS)
Common-mode Rejection	> 40 dB at 60Hz
Input Impedance	10K ohm

ANALOG LINE OUTPUTS

Type	Balanced, low-noise, 2-pole low-pass differential filter
Level (software selectable)	Professional: +4dBu nominal, 20dBu (balanced)
	Consumer: -10dBV nominal, 6dBV maximum (unbalanced)
Frequency Response	+0.0/-0.8 dB, 20 Hz - 20 kHz
THD + N	-98 dB (.0006%) 1kHz signal at -1dBFS
SNR	112 dB (A-weighted)
Dynamic Range	112 dB (A-weighted)
Stereo Crosstalk	< -120 dB, 1kHz
Output Impedance	560 ohms

Specifications: 1820 System

MIC PREAMP/LINE INPUT

Type	TFPro™ combination microphone preamp and line input
Frequency Response	+0.8/-0.1 dB, 20 Hz - 20kHz
Stereo Crosstalk	< 120 dB, 1kHz

LINE INPUT

Gain Range:	-12 to +28 dB
Max Level:	-17 dbV (19.2 dBu)
THD+N:	-94 dB (.002%), 1 kHz at -1 dBFS
Dynamic Range:	100 dB (A-weighted, min. gain)
SNR:	100 dB (A-weighted, min. gain)
Input Impedance:	10K ohm
CMRR:	> 40 dB (60Hz)

MICROPHONE PREAMP

Gain Range:	-10 to +50 dB
Max Level:	-12 dbV (-9.8 dBu)
THD+N:	-95 dB (.0018%), 1 kHz at -1 dBFS
SNR:	100 dB (A-weighted, min. gain)
Input Impedance:	330 ohm
CMRR:	> 80 dB (60Hz)

HEADPHONES

Frequency Response:	+0.0/-0.35 dB, 20 Hz - 20 kHz
THD+N: (1 kHz, max. level)	33 ohm load: -70 dB (0.032%) 600 ohm load: -85 dB (0.006%)
SNR:	112 dB (A-weighted)
Dynamic Range:	112 dB (A-weighted)
Stereo Crosstalk:	< -100 dB (1kHz at -1 dBFS, 600 ohm load)
Max Output Power:	500 mW
Output Impedance:	22 ohms
Gain Range:	85 dB

Specifications: 1820 System

TURNTABLE INPUT	RIAA equalized phono input
Frequency Response:	+/-0.5 dB, 50 Hz - 20 kHz
THD+N:	-76 dB (.015%) (1 kHz, 10 mV RMS unbalanced input)
SNR:	90 dB (1kHz, 10 mV RMS unbalanced input)
Stereo Crosstalk:	< -80 dB (1kHz at -1 dBFS)
Maximum Level:	Professional: 80 mV RMS Consumer: 20 mV RMS
Input Capacitance:	220 pF
Input Impedance:	47K ohm

DIGITAL I/O	
S/PDIF	• 2 in/2 out coaxial (transformer coupled) • 2 in/3 out optical (software switchable with ADAT) • AES/EBU or S/PDIF (switchable under software control)
ADAT	• 8 channels, 24-bit @ 44.1/48 kHz • 4 channels, 24-bit @ 96 kHz • 2 channels, 24-bit @ 192 kHz
Firewire	400 IEEE 1394a port (6-pin) Compatible with DV cameras or HDs
MIDI	2 MIDI in, 2 MIDI out

SYNCHRONIZATION	
Internal Crystal Sync:	44.1kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz ADAT, S/PDIF (optical or coaxial)
RMS JITTER @ 44.1K (Measured via Audio Precision 2)	SRSync SourceRMS jitter in picoseconds 44.1kHz Internal Crystal 596ps 44.1 kHz Optical Input 795ps

Specifications: 1212 System

GENERAL

Sample Rates	44.1 kHz. 48 kHz, 88.2 kHz, 96 kHz, 176.4kHz, 192 kHz from internal crystal. Externally supplied clock from S/PDIF, ADAT (or word clock with optional Sync Card)
Bit Depth	16 or 24-bits
Hardware DSP	100MIPs custom audio DSP. PCI Bus-Mastering DMA subsystem reduces CPU usage. Zero-latency direct hardware monitoring with effects 1394 Firewire Core - Texas Instruments
Converters & OpAmps	ADC - AK5394 (AKM) DAC - CS4398 (Cirrus Logic) OpAmp - NJM2068M (JRC)
WDM Drivers	8 channels — operational at 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz & 192kHz
AudioDockM Power Use	1.25A @ +12V 15W.

ANALOG LINE INPUTS

Type	Servo-balanced, DC-coupled, low-noise input circuitry
Level (software selectable)	Professional: +4 dBu nominal, 20 dBu maximum (balanced)
	Consumer: -10 dBV nominal, 6 dBV maximum (unbalanced)
Frequency Response	+/- .05dB, 20 Hz - 20 kHz
THD + N	-110 dB (.0003%) 1kHz at -1 dBFS
SNR	120 dB (A-weighted)
Dynamic Range	120 dB (A-weighted)
Channel Crosstalk	< -115 dB, (1 kHz signal at -1 dBFS)
Common-mode Rejection	> 40 dB at 60Hz
Input Impedance	10K ohm

ANALOG LINE OUTPUTS

Type	Balanced, low-noise, 2-pole low-pass differential filter
Level (software selectable)	Professional: +4dBu nominal, 20dBu (balanced)
	Consumer: -10dBV nominal, 6dBV maximum (unbalanced)
Frequency Response	+0.0/-0.35 dB, 20 Hz - 20 kHz
THD + N	-105 dB (.0006%) 1kHz signal at -1dBFS
SNR	120 dB (A-weighted)
Dynamic Range	120 dB (A-weighted)
Stereo Crosstalk	< -120 dB, 1kHz
Output Impedance	560 ohms

Specifications: 1212 System

DIGITAL I/O

S/PDIF	• 2 in/2 out coaxial (transformer coupled) • 2 in/3 out optical (software switchable with ADAT) • AES/EBU or S/PDIF (switchable under software control)
ADAT	• 8 channels, 24-bit @ 44.1/48 kHz • 4 channels, 24-bit @ 96 kHz • 2 channels, 24-bit @ 192 kHz
Firewire	400 IEEE 1394a port (6-pin) Compatible with DV cameras or HDs
MIDI	1 MIDI in, 1 MIDI out (16 channels)

SYNCHRONIZATION

Internal Crystal Sync:	44.1kHz, 48 kHz, 88 kHz, 96 kHz, 176.4 kHz, 192 kHz
	ADAT, S/PDIF (optical or coaxial)
	Word Clock (sync card only) - (75 ohm termination, switchable)
RMS JITTER @ 44.1K (Measured via Audio Precision 2)	SRSync SourceRMS jitter in picoseconds 44.1 kHz Internal Crystal 596ps 44.1 kHz Optical Input 795ps

Dimensions & Weight

AUDIODOCK

Complete Product Weight:	5.65lb / 2.56kg
AudioDock Weight Alone:	2.95lb / 1.34kg
Dimensions:	**W**: 8.6" **H**: 1.7" **L**: 9.6" **W**: 218.5mm **H**: 43.2mm **L**: 244mm

1010 PCI Card

Weight:	0.30lb / 0.14kg
Dimensions:	**L**: 6.7" / 170.2mm

0202 Daughter Card

Weight:	0.25lb / 0.10kg
Dimensions:	**L**: 5.04" / 128mm

Internet References

The internet contains vast resources for the computer musician. A few useful sites are listed here, but there are plenty more. Check it out.

Software Updates, Tips & Tutorials................http://www.emu.com

Setting up a PC for Digital Audio ....http://www.musicxp.net

MIDI Basics...Search for "MIDI Basics" (many sites)

MIDI & Audio Recording ....http://www.midiworld.com

MIDI & Audio Recording ....http://www.synthzone.com

ASIO, Cubase & Digital Audio........................http://www.steinberg.net

Cubase Users Group ....http://www.groups.yahoo.com/group/
cubase/messages

Forums

Unofficial E-MU Forum ....http://www.productionforums.com/emu/

E-MU Newsgroup (Yahoo) ....http://groups.yahoo.com/group/e-mu_1820/

KVR Forum...http://www.kvr-vst.com/forum/search.php

Driver Heaven Forum ....http://www.driverheaven.net/search.php?s

MIDI Addict Forum..................................http://forum.midiaddict.com/search.php

Home Recording Forum..........................http://homerecording.com/bbs/
search.php?s=d866b60193933eb726660e7bd
90dfb27

Sound-On-Sound Forum.........................http://sound-on-sound2.infopop.net/2/
OpenTopic?a=srchf&s=215094572

Studio-Central Cafe Forum.....................http://studio-central.com/phpbb/search.php

Sound Card Benchmarking ....http://audio.rightmark.org

Note concerning the Microsoft GS Wavetable Software Synth

Use of the Microsoft GS Wavetable Software Synth will allow only 30 of the 32 ASIO playback channels to work properly. If your PatchMix session uses all 32 ASIO playback channels, you should disable the Microsoft SW Synth before using Cubase LE.

1. Go to: **Start Menu, Programs, Steinberg, Setup MME.**

2. Select **Microsoft GS Wavetable SW Synth** from the MME Outputs.

3. Click the **Set Inactive** button.

Declaration of Conformity

Trade Name:	E-MU Systems
Model No.:	EM8810 & EM8820 EM8810, EM8830 & EM8840 EM8810, EM8830 & EM8841
Responsible Party:	E-MU Systems
Address:	1500 Green Hills Road, Scotts Valley, CA 95066 U.S.A.

This device complies with Part 15 of the FCC rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

CAUTION

You are cautioned that any changes or modifications not expressly approved in this manual could void your authority to operate this equipment.

Note:

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.

- Increase the separation between the equipment and receiver.

- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

- Consult the dealer or an experienced radio/TV technician for help.

The supplied interface cables must be used with the equipment in order to comply with the limits for a digital device pursuant to Subpart B of Part 15 of FCC Rules.

Compliance Information

United States Compliance Information

FCC Part 15 Subpart B Class B using:

CISPR 22 (1997) Class B

ANSI C63.4 (1992) method

FCC Site No.90479

Canada Compliance Information

ICES-0003 Class B using:

CISPR 22 (1997) Class B

ANSI C63.4 (1992) method

Industry of Canada File No.IC 3171-B

European Union Compliance Information

EN55024 (1998)

EN55022 (1998) Class B

EN61000-3-2 (2001)

EN61000-3-3 (1995 w/A1:98)

Australia/New Zealand Compliance Information

AS/NZS 3548(1995 w/A1 & A2:97) Class B

EN55022 (1998) Class B

Japan Compliance Information

VCCI (April 2000) Class B using:

CISPR 22(1997) Class B

VCCI Acceptance Nos. R-1233 & C-1297

Attention for the Customers in Europe

This product has been tested and found compliant with the limits set out in the EMC
Directive for using connection cables shorter than 3 meters (9.8 feet).

Notice

If static electricity or electromagnetism causes data transfer to discontinue midway
(fail), restart the application or disconnect and connect the Firewire cable again.

Index

Notes