

The logo for E-MU, consisting of three horizontal bars on the left followed by the letters 'E-MU' in a bold, sans-serif font.The number '0404' in a large, white, bold, sans-serif font, set against a black rectangular background.The words 'Digital Audio System' in a bold, black, sans-serif font.The words 'Owner's Manual' in a bold, black, sans-serif font.The Creative Professional logo, featuring the word 'CREATIVE' in a bold, sans-serif font above the word 'PROFESSIONAL' in a smaller, bold, sans-serif font, both contained within a black rectangular border.

E-MU 0404 Digital Audio System

Owner's Manual

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Software Version: 1.6

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1- Introduction

Welcome!

Thank you for purchasing the E-MU 0404 digital audio system. Your computer is about to be transformed into a professional quality audio processing workstation. We've designed your E-MU Digital Audio System to be logical, intuitive and above all, to provide you with pristine sound quality. This system offers unprecedented value by providing studio-quality, 24-bit/96kHz multi-channel recording and playback at an astounding price.

E-MU Digital Audio System Components

E-MU 0404	Inputs & Outputs
<ul style="list-style-type: none"> ▪ E-MU 0404 PCI Card ▪ Analog Breakout Cable (1/4") ▪ Digital Breakout Cable ▪ Install & Application CDs ▪ This Owner's Manual ▪ Quick Start Guide 	(2) Ch. S/PDIF Optical In (2) Ch. S/PDIF Optical Out or (2) Ch. S/PDIF Coaxial In (2) Ch. S/PDIF Coaxial Out (1) MIDI Input & Output (16 ch.) (2) 24-bit unbalanced Line Inputs (2) 24-bit unbalanced Line Outputs

The System Includes:

The E-MU 0404 PCI Card provides 2 line level, unbalanced analog inputs, 2 line level, unbalanced analog outputs, plus MIDI input and output. This is a finely-tuned audio interface, using high performance 24-bit/96kHz A/D - D/A converters to deliver an unbelievable 111dB of dynamic range. Check out the complete specs on [page 96](#).

The PCI card contains a powerful hardware DSP processor which allows you to use over 16 simultaneous hardware-based effects, which place minimal load on your computer's CPU. The E-MU 0404 PCI Card also provides a S/PDIF stereo digital input and output with either optical or coaxial connections. A built-in MIDI interface allows you to connect external MIDI instruments or keyboards directly to your computer.

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) inputs and outputs and internal hardware effects and buses—no external mixer needed. You can add digital effects, EQs, meters, level controls and ASIO 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, special effect setups or general computer use.

Optional Sync Daughter Card

The optional Sync Daughter Card adds Word Clock in and out for sample-synchronizing outboard digital equipment, SMPTE longitudinal time code in/out for syncing other recording equipment and 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

Prior to installing the hardware, take a few moments to enter the 18-digit serial number of the E-MU 0404 PCI Card. This number can help EMU Customer Service troubleshoot any problems you may encounter—by writing the numbers down now, you'll avoid having to open your computer to find it later on.

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 0404 PCI card in your computer. [Go there.](#)
3. Install the optional Sync Card (if you have one). [Go there.](#)
4. Attach the Analog and Digital breakout cables to the rear of the 0404 card.
5. Install the PatchMix DSP software onto your computer.
6. Connect audio, MIDI and synchronization cables between the Digital Audio 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 currently not supported.

Please read the following sections as they apply to your system as you install the E-MU 0404, paying special attention to the various warnings they include.

Safety First!

- To avoid possible permanent damage to your hardware, make sure that all connections are made to the E-MU 0404 PCI card 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 0404 PCI board and the interfaces are susceptible to electrostatic discharge, commonly known as "static." Electrostatic discharge can damage or destroy electronic devices. Follow these procedures 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.

▼ As you install hardware components, observe the following general precautions to avoid damage to your equipment and yourself.

Connector Types

These connector types are used to connect the E-MU 0404 hardware components. They will be referred to by the name shown in the first column of the following chart:

Name	Description	Connects
DB-15 Digital	Digital Cable Connector	0404 PCI card and Digital I/O
DB-9 Analog	Analog Cable Connector	0404 PCI card and Analog I/O
1/4" Jacks	1/4" Breakout Cable	2 channel analog input/output
S/PDIF In	RCA Connector	S/PDIF digital audio devices
S/PDIF Out	RCA Connector	S/PDIF digital audio devices
S/PDIF Optical In	TOSLINK Optical Connector	S/PDIF digital audio devices
S/PDIF Optical Out	TOSLINK Optical Connector	S/PDIF digital audio devices

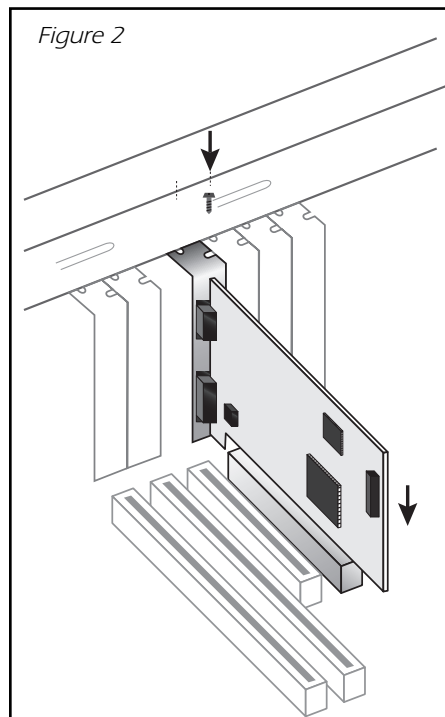
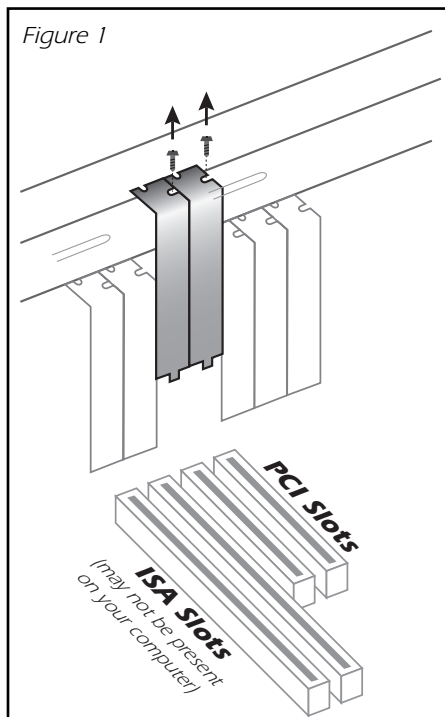
Installing the E-MU 0404 PCI Card

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 0404 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 bracket from any available PCI slot. If you are installing a Sync Card as well, remove the bracket from two adjacent PCI slots. See figure 1 below.

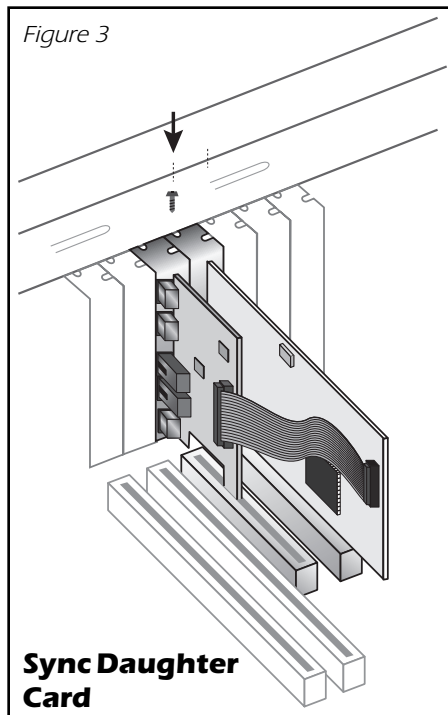
■ **Note:** Some computer cases don't use screws to secure PCI cards. In this case, follow the instructions that came with your computer.



5. Align the E-MU 0404 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 0404 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.
8. Attach the Analog and Digital breakout cables to the rear of the 0404 card.
9. Connect your audio cables to the breakout cables.

Install the Sync Card

1. Unwrap the Sync Card if you have one and get ready to install it.
2. Connect the ribbon cable provided with the kit between the E-MU 0404 card and the Sync Daughter Card as shown in figure 3. The cable is keyed so it cannot be incorrectly inserted. Seat the connectors firmly in the sockets and arrange the cable neatly.
3. Align the Sync Card with the slot and press gently but firmly down into the slot as shown in figure 2 on the preceding page.
4. Do not force the Sync Card into the slot. Make sure that the 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.
5. Secure the card into the slot using one of the screws you placed aside earlier.



Software Installation

Installing the E-MU 0404 Drivers

The first time you restart your PC after installing the E-MU 0404 PCI card, you will need to install the PatchMix DSP software and E-MU 0404 PCI card drivers.

Windows 2000 or Windows XP

The software is not compatible with other versions of Windows.

1. After you have installed your audio card, turn on your computer. Windows automatically detects your audio card 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:\ctrun\ctrun.exe** (replace **d:** with the drive letter of your CD-ROM drive). You can also open the CD and double-click **Ctrun.exe** located in the **CTRun** folder.
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 note below for more information.
6. When prompted, restart your computer.

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.

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 0404 PCI card entry, or the application entry 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 0404 PCI card device drivers or applications.

■ Serial Number -

During the registration process, you will be asked to enter your serial number. The serial number is located on the back of the box and on the 0404 PCI Card.

3 - PCI Card & Interfaces

The E-MU 0404 PCI Card

The E-MU 0404 PCI card contains E-MU's powerful E-DSP chip which leaves more power free on your CPU for additional software plug-ins and other tasks. Bit depth is controlled by your recording or audio application. The 0404 PCI card always sends and receives 24-bit audio.

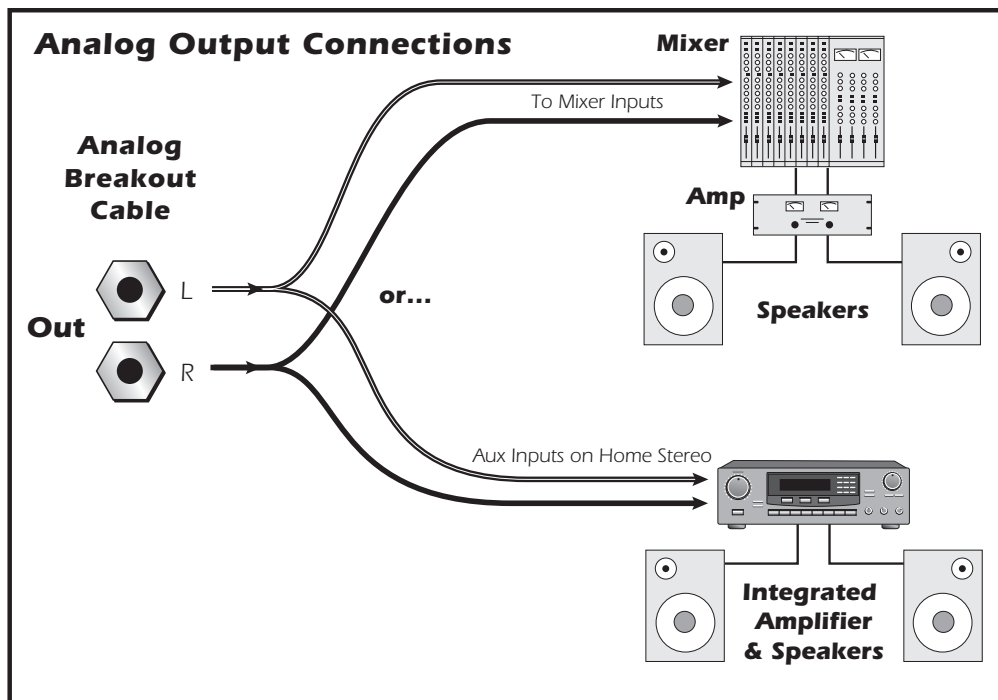
DB-9 & DB-15 Connectors

Connects the analog and digital breakout cables to the 0404 PCI card. Connect the 1/4" Analog Breakout Cable to the DB-9 connector and the Digital Breakout Cable to the DB-15 connector.

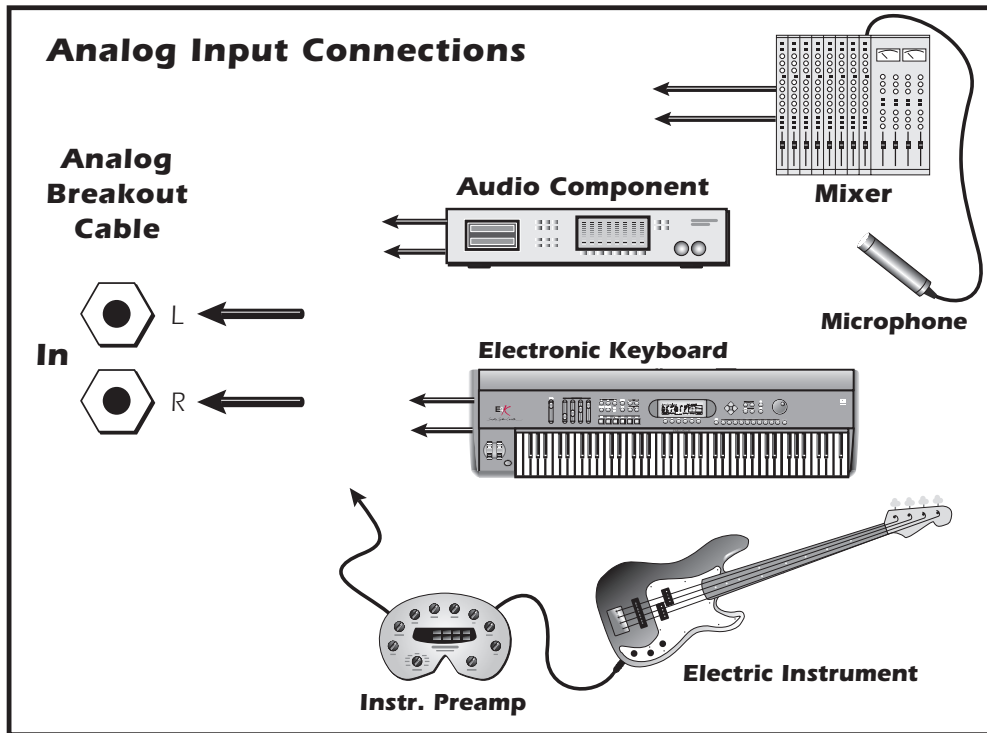
Analog Breakout Cable

The 0404 PCI card provides one pair of 24-bit unbalanced analog inputs and one pair of 24-bit unbalanced analog outputs. The analog breakout cable is designed to accommodate 1/4" plugs. Use 1/4" to RCA adapter cables to connect to consumer audio gear.

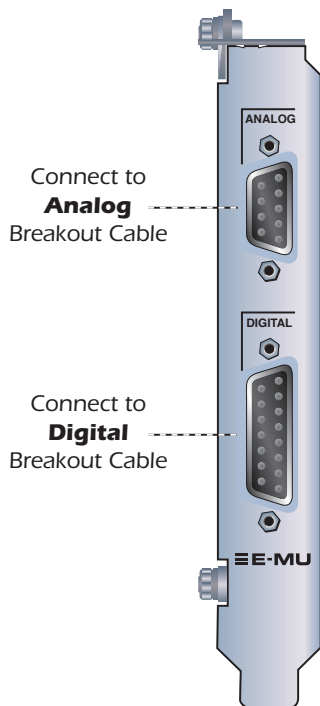
The outputs can feed any line level input such as a mixing board, the auxiliary input on your stereo, or a set of powered speakers. They are not designed to drive headphones directly. Use a mixer, home stereo receiver, or headphone amplifier to monitor with headphones.



The inputs can be connected to any line level stereo signal from keyboards, CD-players, cassette decks, etc. Use 1/4" to RCA adapter cables to connect to consumer audio gear.



Digital Breakout Cable



S/PDIF Digital Audio Input & Output

RCA phono jacks are the standard coaxial connectors used for S/PDIF (Sony/Philips Digital InterFace) connections. A single jack carries two channels of digital audio. The E-MU 0404 receives digital audio data with word lengths of up to 24-bits. Data is always transmitted at 24-bits.

S/PDIF digital I/O allows you to receive and/ or transmit of digital data from external digital devices such as a DAT, external analog-to-digital converters or external signal processors equipped with digital inputs and outputs.

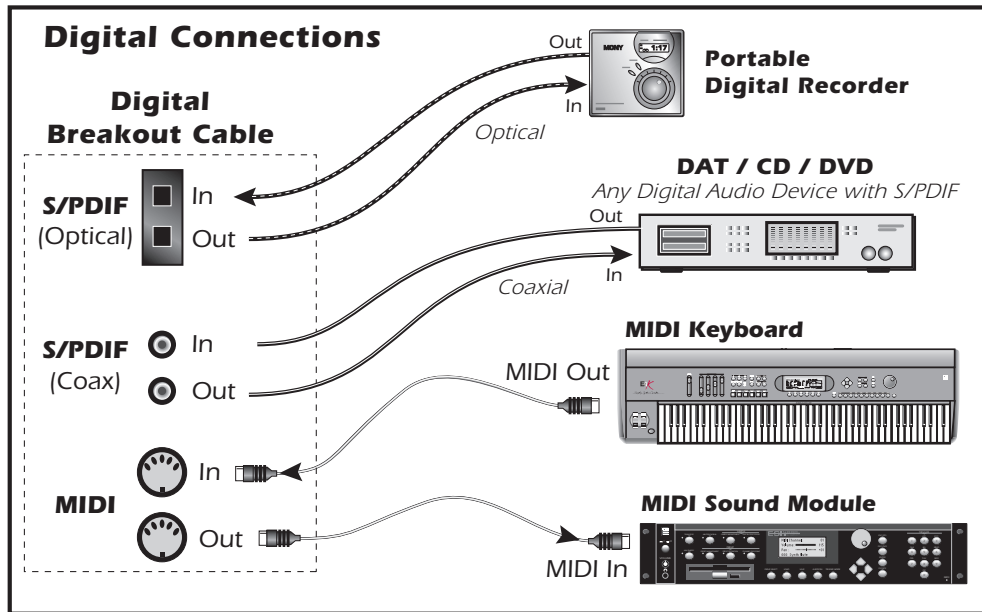
S/PDIF can also be transmitted and received via the TOSLINK optical connectors on the Digital Breakout Cable. Optical connections have certain advantages such as immunity to electrical interference and ground loops. Make sure to use high quality glass fiber light pipes for connections longer than 1.5 meters.

The optical S/PDIF and RCA coaxial S/PDIF Inputs cannot be used simultaneously, however BOTH S/PDIF Outputs are available simultaneously (carrying the same signal). [See System Settings.](#)

The S/PDIF out can be configured as either Professional or Consumer mode in the **Session Settings** menu. The 0404 PCI card can be connected to AES/EBU digital audio systems 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 and 96kHz sample rates. The word clock contained in the input data stream can be used as a word clock source. [See Using External Clock](#).

▼ **Important:** When using any type of digital I/O such as S/PDIF, you **MUST** sample sync the two devices or clicks and pops in the audio will result. [See Using External Clock](#)

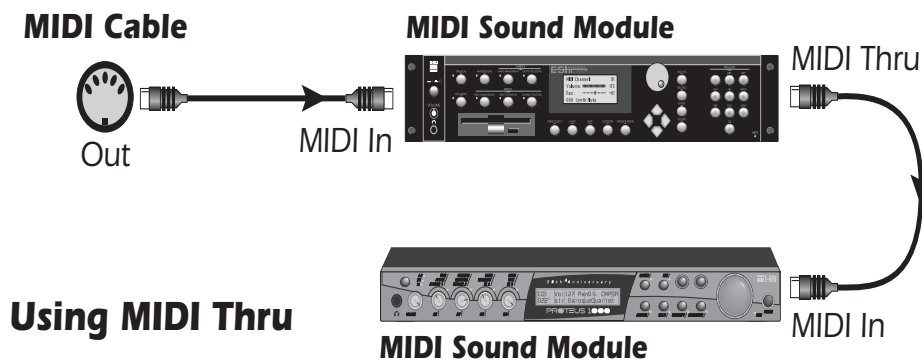


MIDI (Musical Instrument Digital Interface) is a standard specification for networking two or more devices together. Connect **MIDI Out** to external MIDI instruments and connect **MIDI In** to a controller such as a MIDI keyboard.

Unlike S/PDIF, the MIDI cable does not carry audio data. In its most basic application, MIDI tells a synthesizer when to start and stop playing specific notes. MIDI also carries other information such as how hard the note was played, what sound to play, the channel volume and many other commands. The most important thing to remember is that MIDI contains **CONTROL** information, not the sound itself.

Information on the MIDI cable can be assigned to any one of sixteen channels so that different musical lines can be assigned to play specific sounds or MIDI instruments.

To connect more than one MIDI instrument to the 0404 PCI card, the MIDI Thru port on your synthesizer can be used. MIDI Thru carries an exact copy of the data on the synthesizer's MIDI input port. To find more information about MIDI, see [page 91](#).



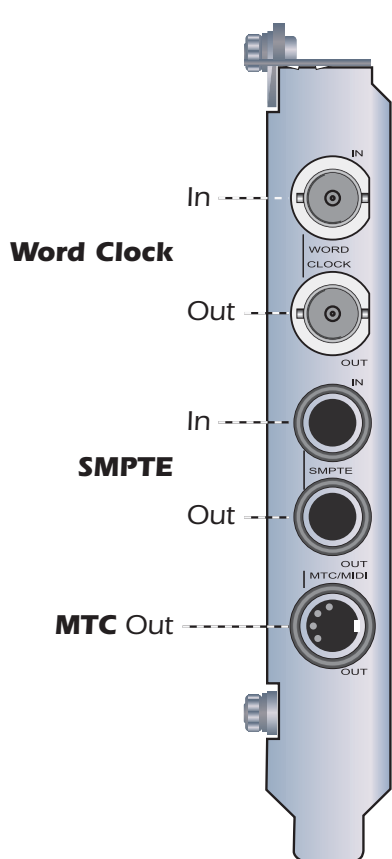
The Sync Daughter Card Option

The Sync Daughter card option adds 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 S/PDIF optical are all industry standard methods of synchronizing digital equipment together at the system sample rate (44.1kHz, 48kHz, or 96kHz). 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, or S/PDIF 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.



Connecting Cables to the Sync Daughter Card

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

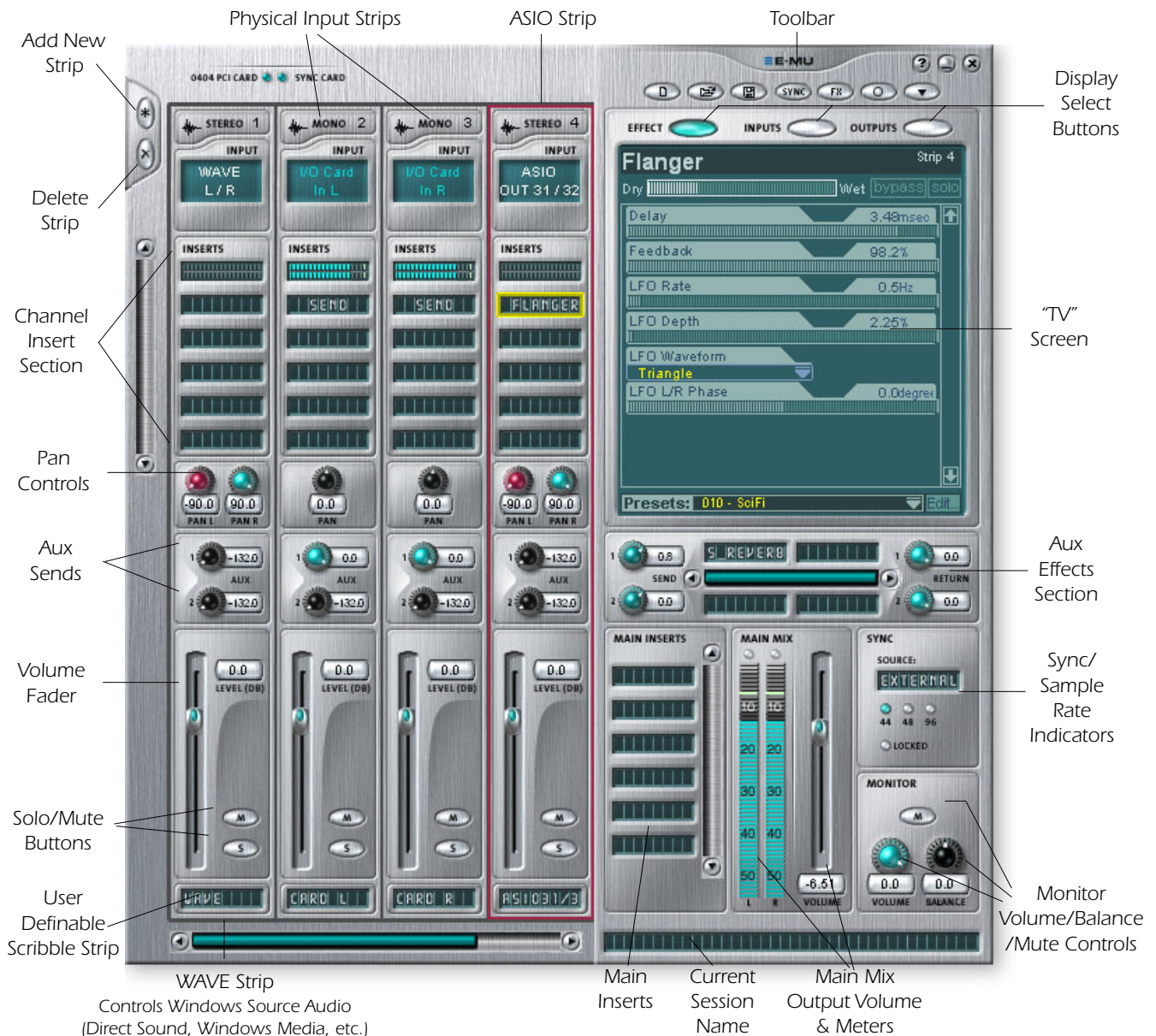
The PatchMix DSP Mixer is a virtual console, which performs all of the functions of a typical hardware mixer and then adds a few new tricks of its own. PatchMix DSP greatly simplifies audio operations such as ASIO/WAVE routing, volume control, stereo panning, equalization, effect processing, effect send/return routing, main mix and monitor control, without getting in the way of your other software. It's easy and it works...beautifully!

► 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.

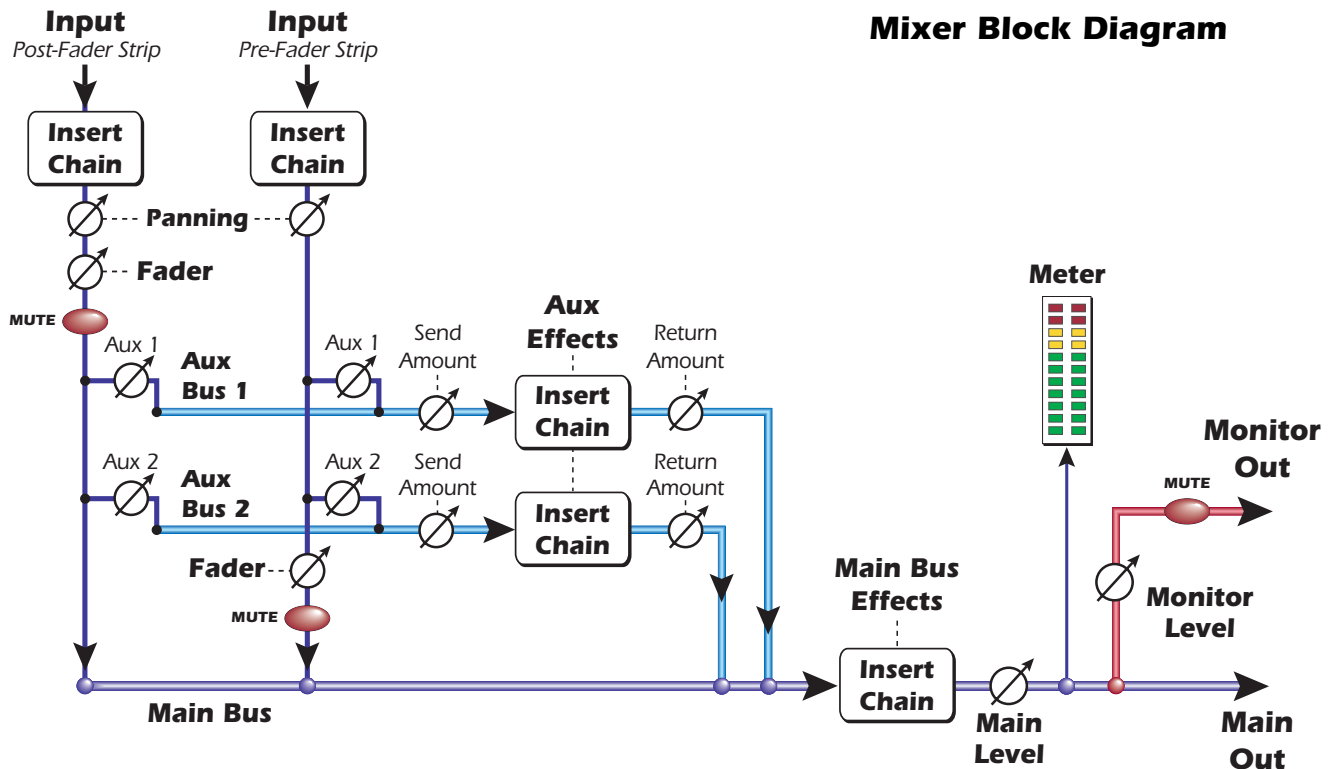
Application Toolbar Lets you manage sessions and show/hide the various views.

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 the Digital Audio System is 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.

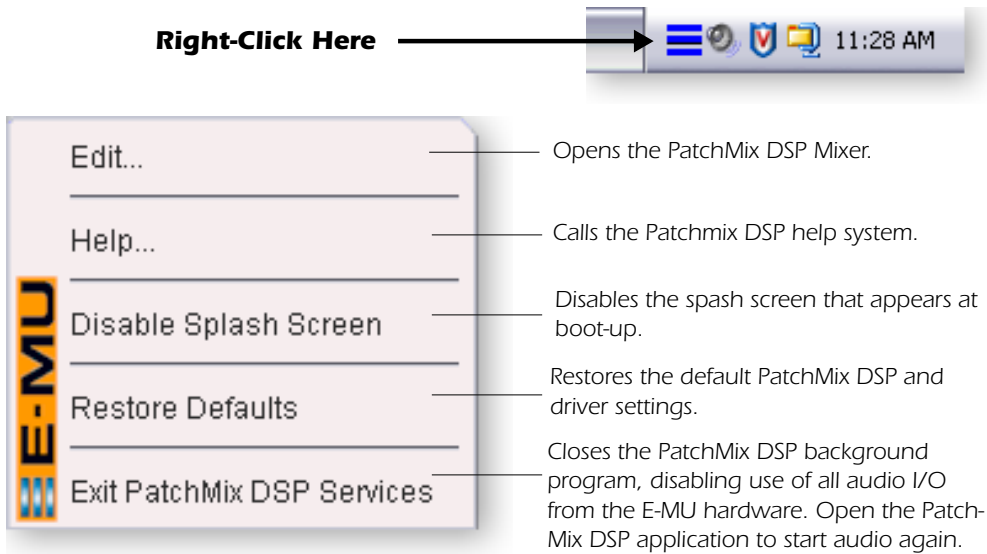


Pre Fader or Post Fader

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.](#)

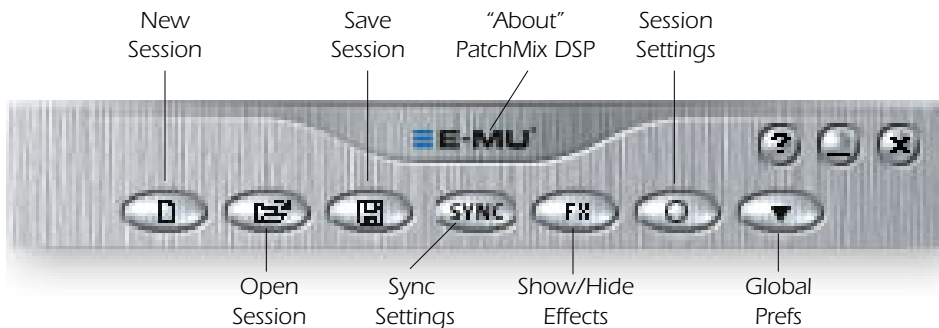
E-MU Icon in the Windows Taskbar

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



❖ Click the buttons in the toolbar to learn about their function.

The Toolbar



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 44.1kHz, 48kHz, or 96kHz sample rates, but the effect processors are only available at the 44.1kHz or 48kHz rates. See [96kHz Operation](#) for complete details.

When you start a new PatchMix DSP Session, the first choice you make is to select the sample rate. Once set, you can only easily switch between 44.1kHz and 48kHz. You cannot switch between 44/48kHz and 96kHz. This is because the number of mixer inputs and outs change significantly at the higher sample rate. 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 either 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, the Sync Indicator will be extinguished (off), but PatchMix will attempt to receive the external data. If the Sync Indicator is Off, the two units are NOT sample locked and you should correct this condition to avoid intermittent clicks in the audio.

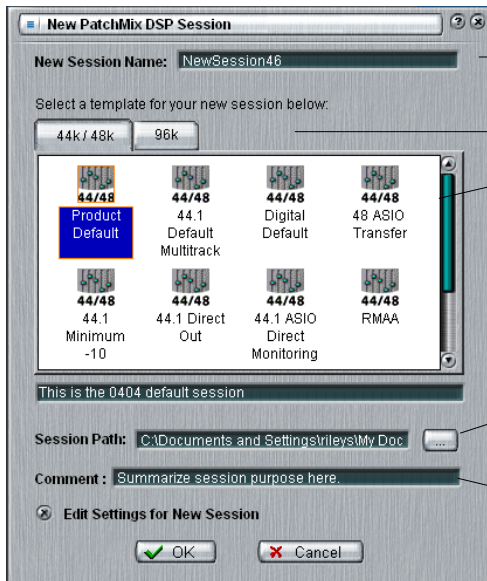
PatchMix DSP comes with several session templates to choose from so when you create a new session you can either create a “blank” session based around a designated sample rate, or select from a list of template starting points. **Always check for the presence of the “LOCKED” indicator whenever you are using a digital interface.**

In a PatchMix DSP session the number of strips in the mixer is dynamically configurable. See [Pre Fader or Post Fader](#). 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. See [Using External 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.



Name the New Session

Select a Template or new Session at the desired sample rate

Choose the location where the new Session will be saved.

Add your own comment or note about the 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 two groups based on sample rate, 44.1kHz/48kHz or 96kHz.

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 (0404) 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.

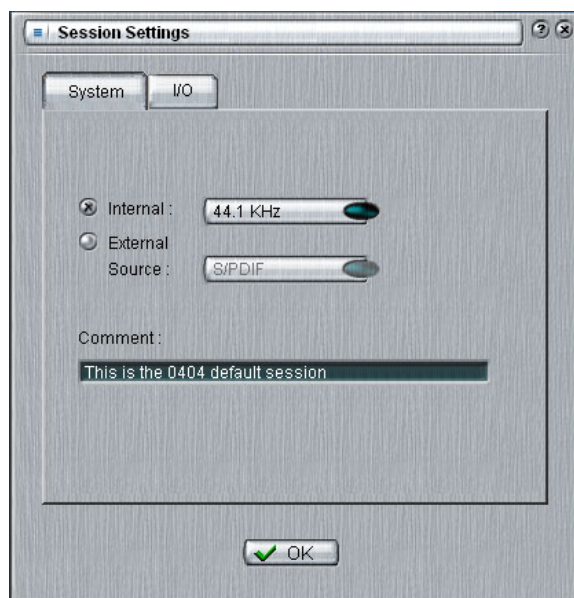
Get in the habit of saving the session whenever you have created a special mixer setup. This will make your life much easier as you can recall a setup for many different audio modes such as: recording, mixing, special ASIO routings, etc.

❖ 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 or I/O options.



The System Settings include the following choices:

▪ Internal/External Clock	Selects between internal or external 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, 96kHz.
▪ External Clock Source (ext. clock only)	Select from S/PDIF or the 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

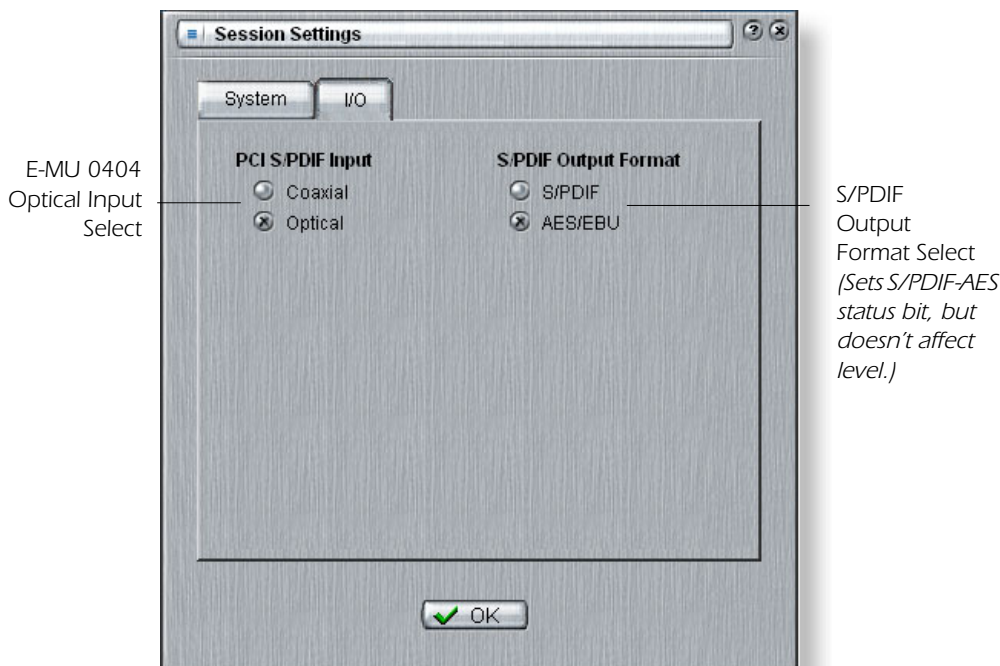
Whenever you are connecting two or more devices using digital I/O such as S/PDIF, one of the digital devices **MUST** supply the master clock to the others. This master clock runs at the system sample rate (44.1, 48, or 96kHz) and can be distributed using a dedicated cable (word clock) or **embedded into a data stream such as S/PDIF**. 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.**

If an External Clock is interrupted or switched after the Session has been created (except between 44.1k <-> 48k), the "LOCKED" indicator will extinguish 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.

I/O Settings

The 0404 PCI card is optimized for signal levels of -10dBV (consumer standard) for the analog inputs and outputs. -10dBV levels are compatible with most consumer audio gear.

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.



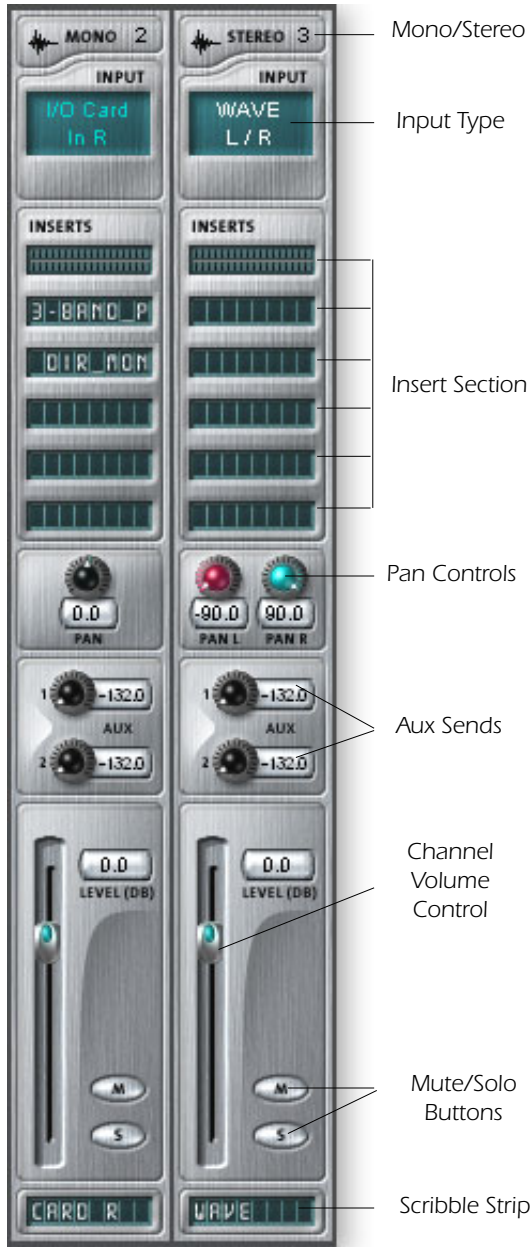
The optical digital TOSLINK input and output on the Digital Breakout Cable can be used to transmit and receive stereo S/PDIF.

▪ PCI Card S/PDIF Input	Selects between coaxial or optical S/PDIF input. S/PDIF out is always transmitted on both the coaxial and optical outputs.
▪ S/PDIF Output 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 Line Input Mixer Strips are mono. The WAVE and S/PDIF strips are stereo. Each input mixer strip can be divided into four basic sections.

▪ Insert Section	Effects, EO, External/Host Sends and Returns can be inserted into the signal path.
▪ Pan Controls	This control positions 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.



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)
- **Host Input**
(Direct Sound, WAV, ASIO source)

❖ The **Input Type** will turn RED if the input is not available.

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

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

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.

❖ To learn how to record monophonic inputs in stereo, see page 72.

Pan Controls

These controls allow you to 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.

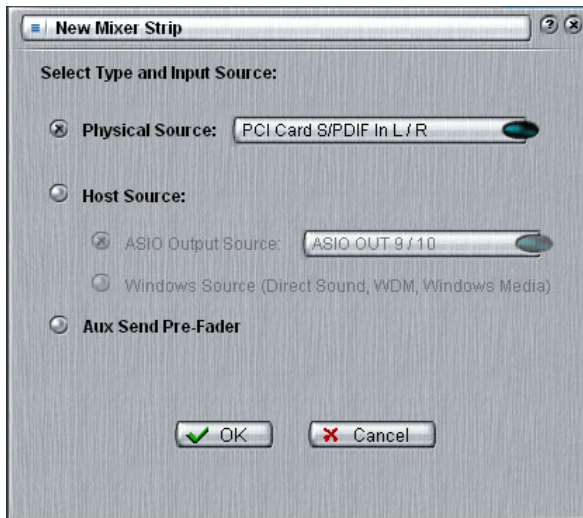
This screen shows a mono strip on the left and a stereo strip on the right.

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. [See Overview of the Mixer](#)
2. The Assign Mixer Strip Input Dialog appears:



3. Select the desired input to the mixer strip from the following choices:

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

❖ Use the **WAVE** strip to playback CDs, Windows Media Player, and non-ASIO audio applications.

Mixer Strip Type	Function
Physical: PCI Card Analog	24-bit monophonic analog input.
Physical: PCI Card S/PDIF	2 channel digital audio from the S/PDIF input.
Host: WAVE L & R	Direct Sound, WDM, Windows Media (Sound generated or handled by Windows, such as game sound, CD player, beep sounds, etc.).
Host: From ASIO Out	2 channel digital audio from an ASIO source (software application).

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.

❖ [See "Pre or Post Fader Aux Sends" on page 29.](#)

► 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. [See Overview of the Mixer](#)

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. Special inserts, 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 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 available in your ASIO software.

The following types of inserts can be selected.

Hardware Effect	Reverb, EQ, Compressor, Flanger, etc. using PatchMix DSPs effects which do not load your CPU.
ASIO Send	Splits off the signal and sends 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” .

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

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.

Note: The Physical Output & Input option is “grayed-out” when using the default Session

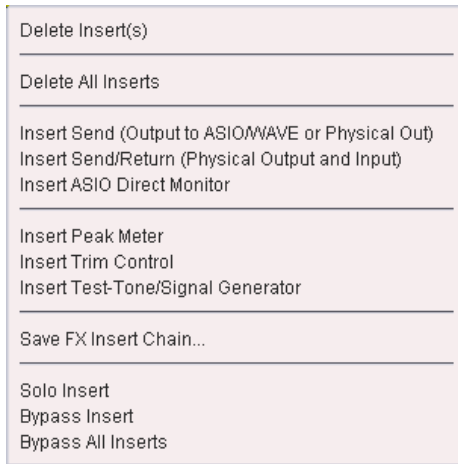
Reason: The 0404 Digital Audio System has only 4 physical inputs and 4 physical outputs. The Send/Return option is grayed-out because all the physical I/O resources available for send/return have been used in this Session. If S/PDIF I/O is not being used elsewhere, it becomes available in the Send/Return list.

▼ You are not able to select analog return when using the Send/Return Inserts.

Reason: The 0404 Digital Audio System only supports **Stereo I/O** in the Send/Return Insert section. The analog inputs on the 0404 are mono.

The Insert Menu

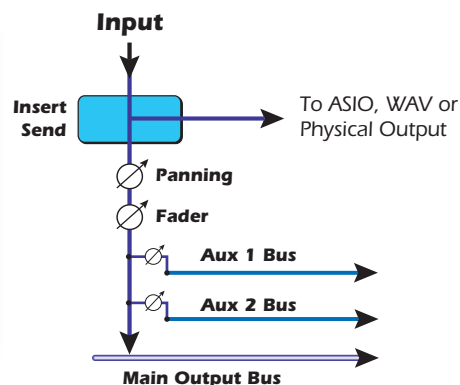
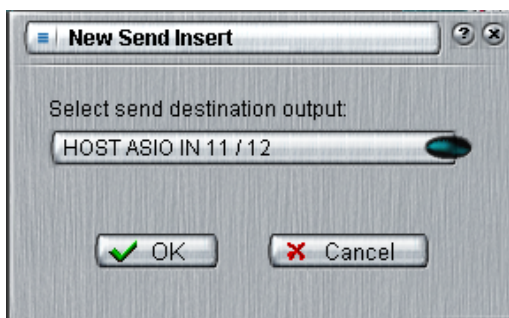
Right-Clicking over the insert section brings up a pop-up selection box containing various insert options to help you control and manage your inserts.



► 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 other)” from the list of options. The following dialog box appears.

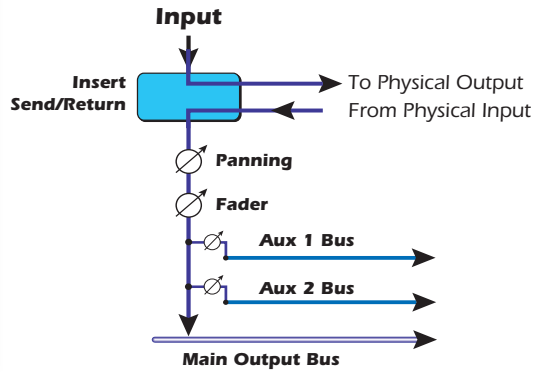
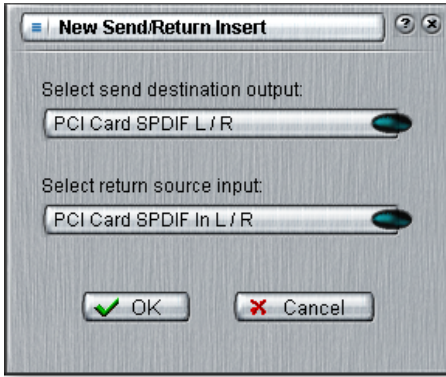


3. Choose one of the Send Outputs. Click on a destination to select it.
4. Click OK to select the output or Cancel to cancel the operation.

► 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” 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.

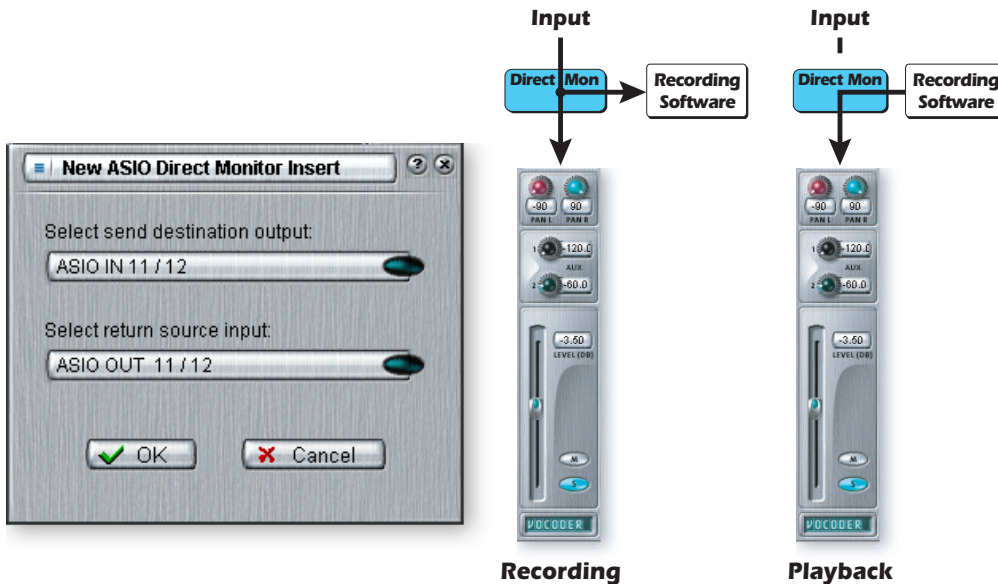
3. Choose one of the Send Outputs. Click on a destination to select it.
4. Choose one of the Return Inputs. Click on a source to select it.
5. Click OK to select the Send and Return or Cancel to cancel the operation.

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 Cubasis). 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.



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

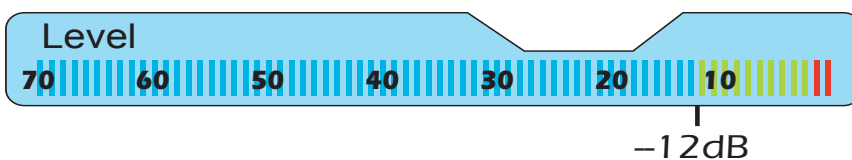
Keeping track of signal levels is important in any audio system, be it analog or digital. You want to keep the signal levels running as close to maximum in order to achieve high resolution and low noise. On the other hand, you don't want the signal level so high as to cause clipping. To help you maintain 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 1 dB. The yellow bars begin at -12dB below full scale.

The insert meters are also useful to monitor incoming digital signals such as 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.

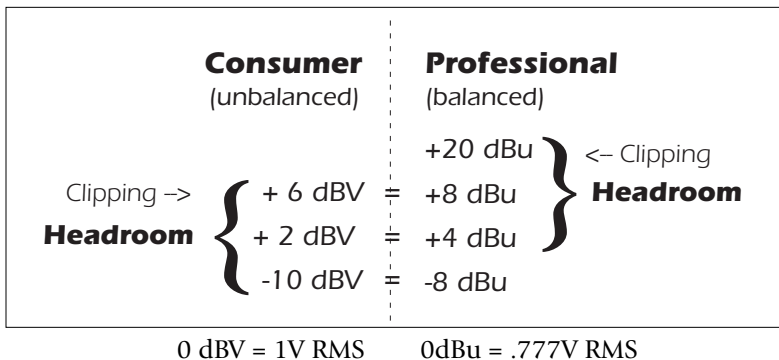
► **To Insert a Meter**

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 FX in the Main Section, then Left-Click on the meter insert. 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 0404 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.

Comparison of -10dBV & +4dBu Signal Levels



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.

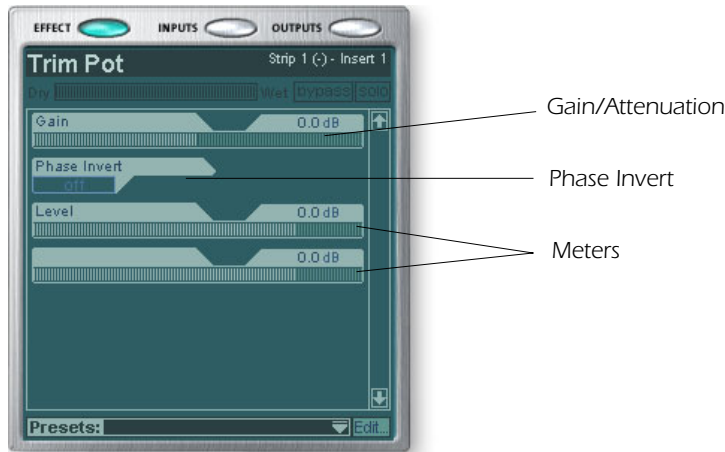
Play your input source signal while watching the insert meter in the strip. The signal should go into the yellow area frequently, but never into the red. Adjust the level of your source until you have a good level. If the signal is way too strong or too weak, you may have to go back and adjust the I/O 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 $\pm 30\text{dB}$ 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 signal send or return from an external effect, or use it to drive an effect device. Certain effects such as the Compressor, Distortion, or 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.

The phase invert switch inverts the polarity of the signal. It is generally used to correct for 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 Effects from an Insert:

1. Right-Click over the Insert Effect 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 Plug-in 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 FX 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 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 FX 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.

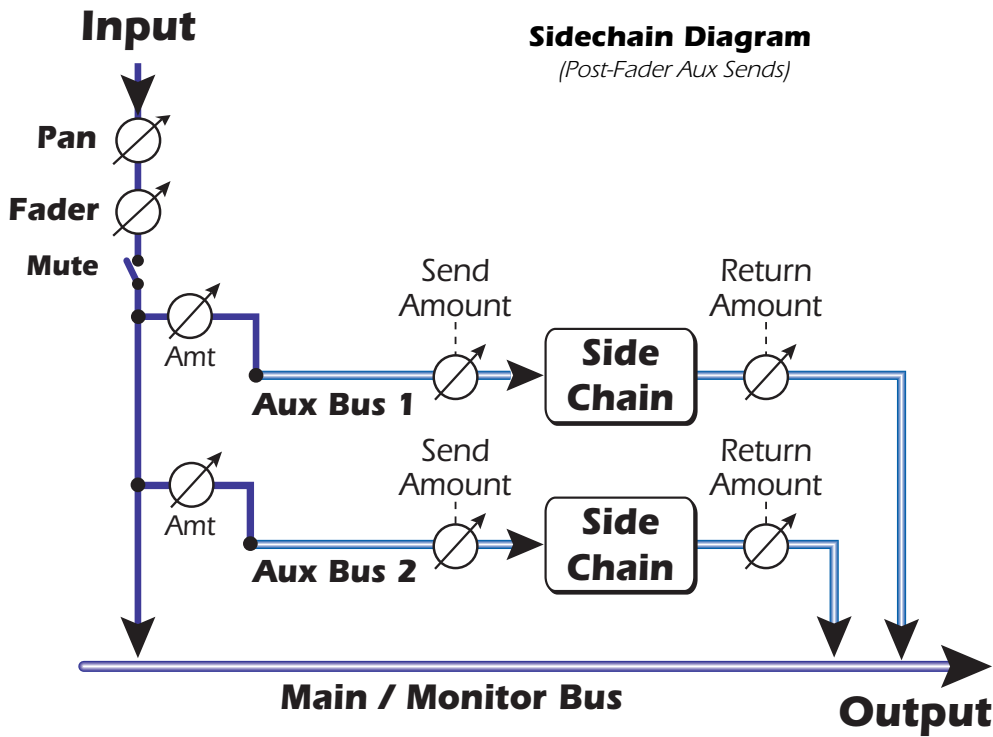
Aux Section

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.



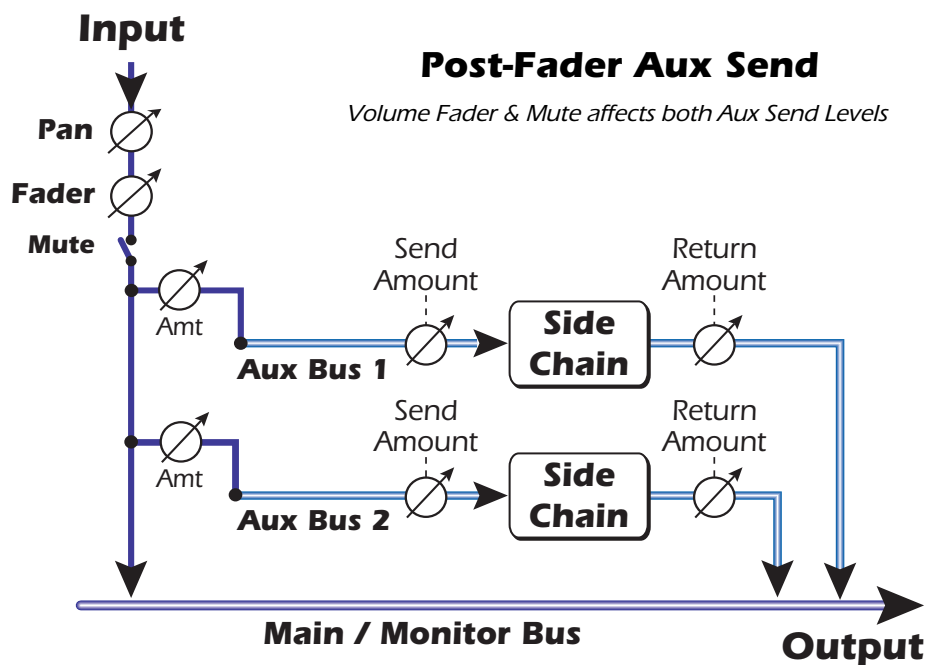
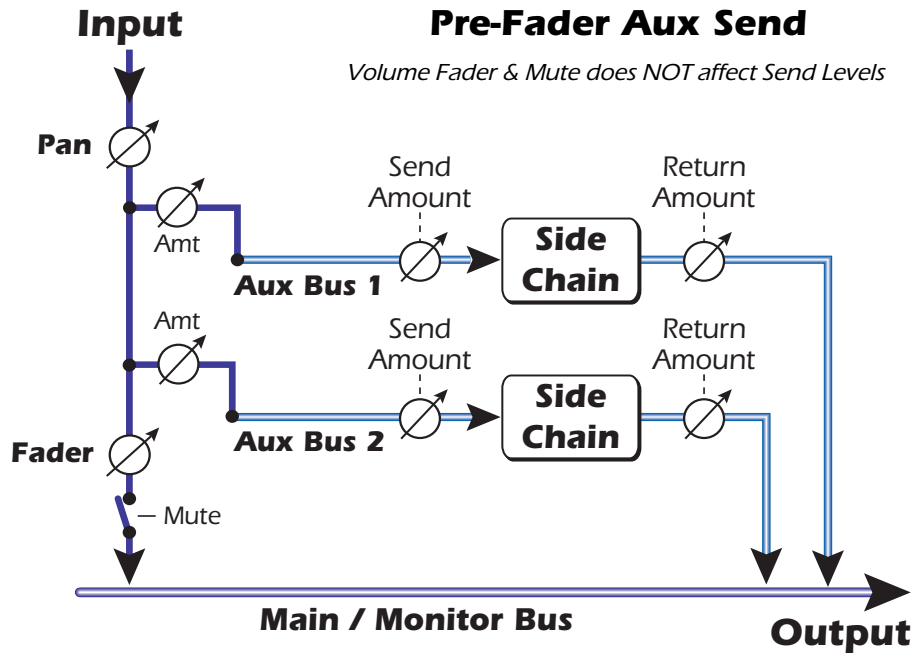
Other Uses of the Aux Sends

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.

Level, Pan, Solo & Mute Controls



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.

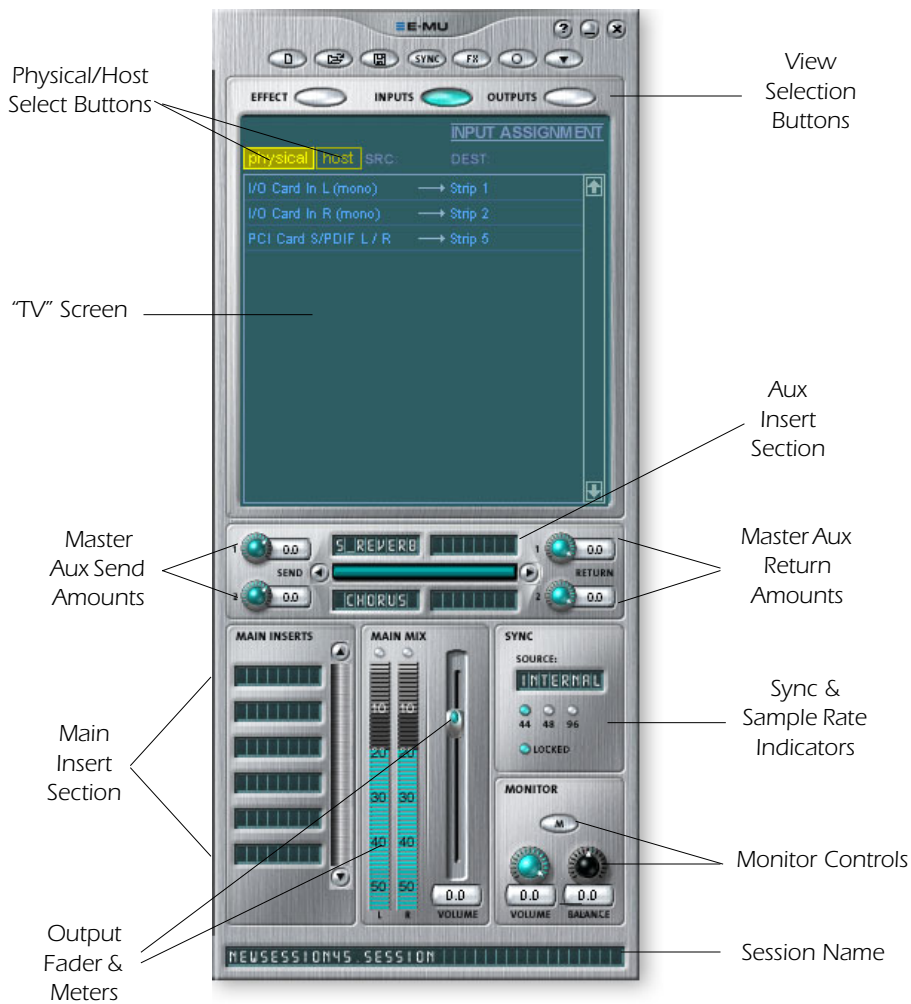
The **Solo** button allows you to listen to only that channel while muting the rest of the mixer’s output. If multiple solo buttons are pressed, you will hear all soloed channels and the non-soloed channels will all be muted.

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 effect 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

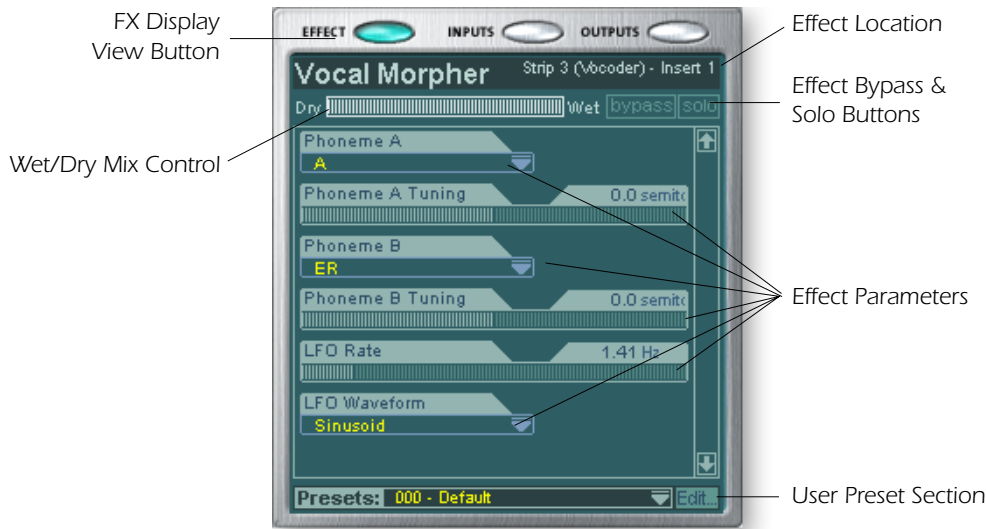
The “TV screen” at the top of the main section is a multi-function display and control center for the input and output routings and effect controls. The three buttons at the top of the display select the current function of the display—Effect, Inputs or Outputs.

Effect

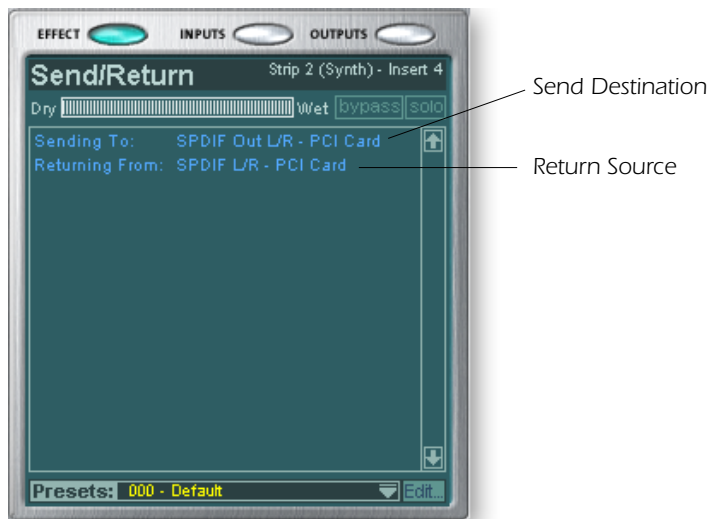
Select the **Effect** display view in the main section, then click on an Effect Insert to display the effect parameters. If an insert effect is not selected, the display will read “No Insert”.

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 FX preset. The parameter set varies with the type of effect. See [“Core Effects Descriptions”](#) 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 FX display enabled, the TV screen shows you where the Send is going and where the Return is coming from. The buttons at the top of the display allow you to bypass or solo the Send/Return insert.



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 screen, 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. Clicking on any of the input routings in the TV display highlights the corresponding mixer strip.

❖ 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

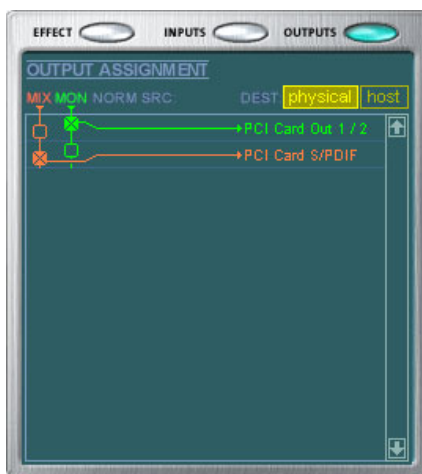


❖ **Tip:** Clicking on any of the input routings in the TV display highlights the corresponding mixer strip.

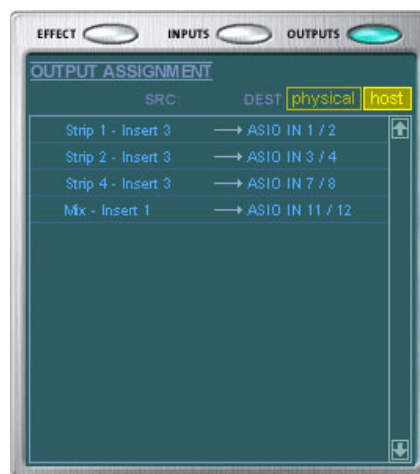
Output

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.

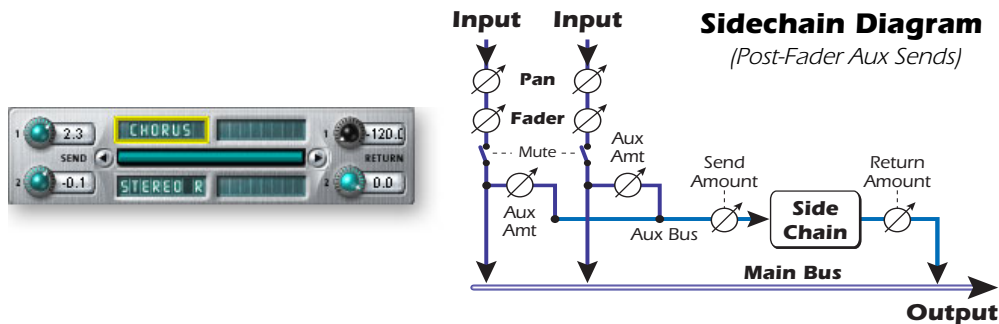
The **Host Output** screen displays and allows you to view the Host (ASIO or WAVE) outputs of the mixer. See [“Insert Section”](#) for information on how to connect the inserts.

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.



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

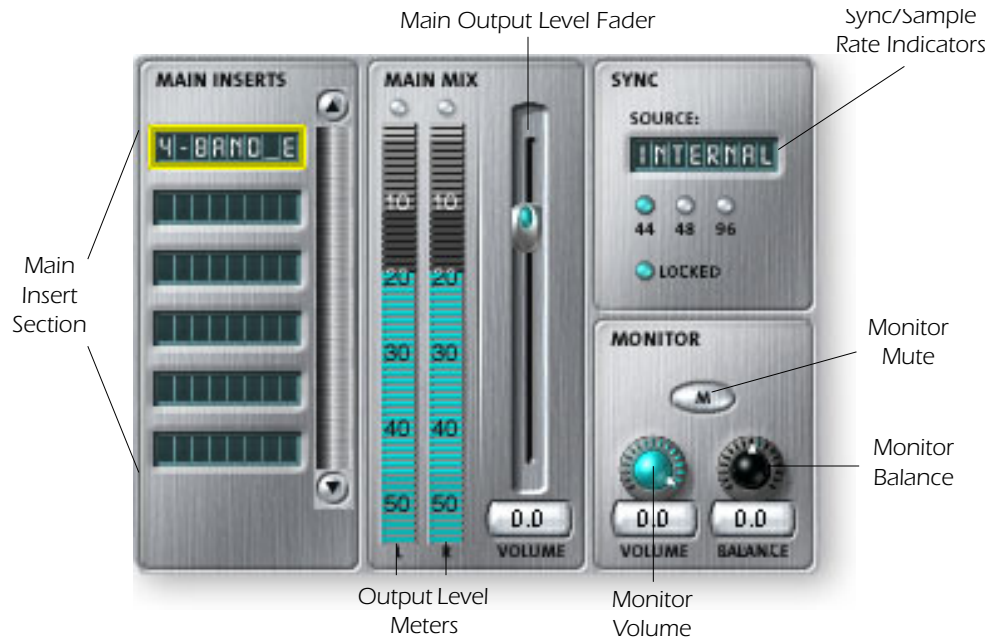


The Sync/Sample rate Indicators show the current session’s sample rate and whether it is internal or slaving to an external source. The LEDs indicate which sample rate is currently in effect. If an external source is being used, the Source display reads “EXTERNAL”.

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, and 96k, 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 of 96k, 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



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 effect 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.**

Monitor Output Level

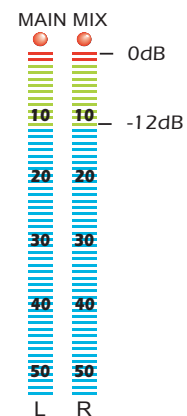
This control adjusts the monitor output level. Keep in mind that since the monitor level control comes after the Main Output Fader, nothing will be heard from your monitors if the main level is turned down.

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.

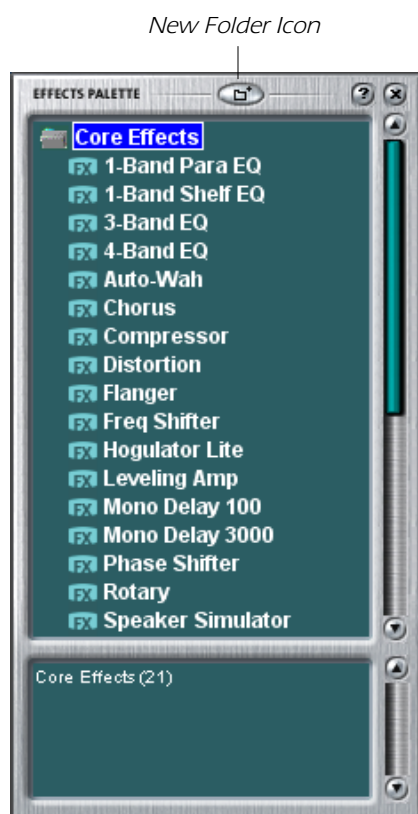
Since the effects are implemented in hardware, they don't place any load on your host computer. This allows your valuable CPU cycles to be used for other applications or software plug-ins. The effects are only available at the 44.1 and 48kHz sample rates.

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 ["FX Insert Chains" on page 38](#).



Effect Categories

Core Effects

Multi-Effects

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

► 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 one or several effects and their settings into a single multi-effect. When an effects chain is selected and placed 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 one or more effects and place them into any insert location in the mixer.
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."
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, warning you that this action will delete all presets in the folder.
2. Select "Delete Category". A popup selection box appears.
3. Click OK to delete the folder or Cancel to cancel the operation.

► To Rename an FX 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.

96kHz Operation

When operating at the 96kHz sample rate, the effect processors are completely disabled. However, the Inserts, Send>Returns, Meters, Trim Controls, Test Tones and ASIO Direct Monitoring ARE fully functional.

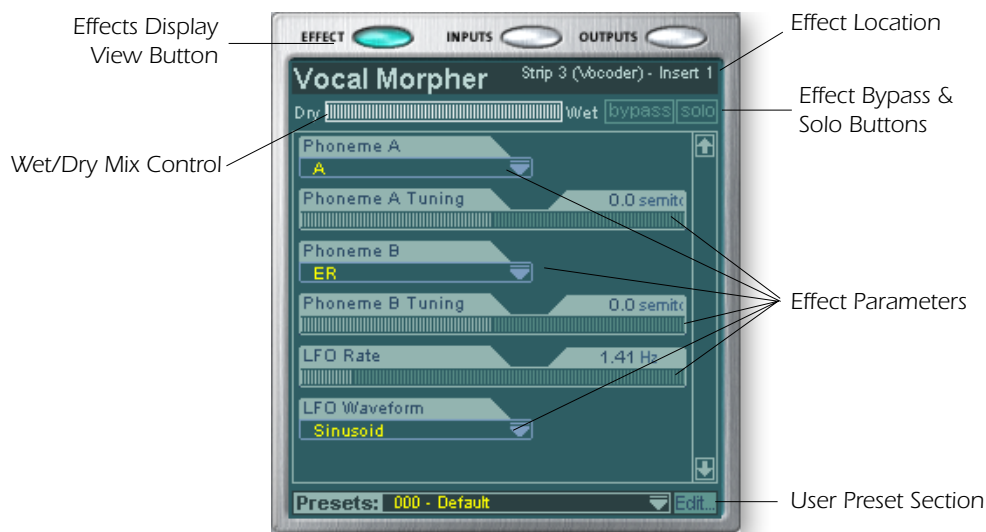
FX Edit Screen

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 Insert (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 effect name will “gray-out” to indicate that the 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 (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 "gray-out" to indicate that they are bypassed.

► To Un-Bypass ALL

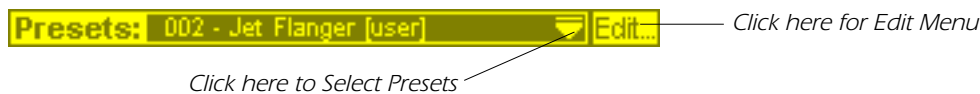
All the inserts in a strip can also be un-bypassed with a single command. This command works even if only some of the effects are bypassed.

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.



► 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\E-MU PatchMix DSP\Effect Presets") can be copied, e-mailed or shared like any other computer file.

Hint: Open with "Notepad" or other word processor to view and edit the name and parameters.

WDM Recording and Playback Behavior

WDM record and playback is now 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, record and playback 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 record or 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 96kHz, 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 recorded or played back. In other words, the WDM driver does not perform sample rate conversion of any kind when PatchMix is running at 96kHz.

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 550
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.

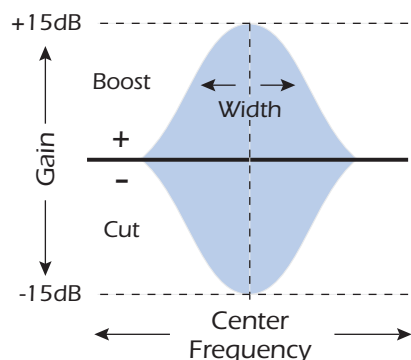
Examples of Effects Usage *(with a WAVE, ASIO Return & 2 Inputs)*

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

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

Core Effects Descriptions

1-Band Para EQ

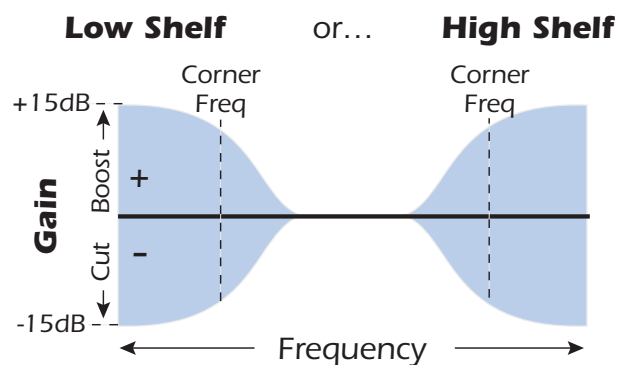


This single band parametric equalizer is useful when you just want to boost or cut a single range of frequencies. For example, if you just want to brighten up the lead vocal a bit, you might choose this EQ. This EQ offers up to ± 15 dB cut or boost.

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: 1 semitone to 36 semitones

1-Band Shelf EQ

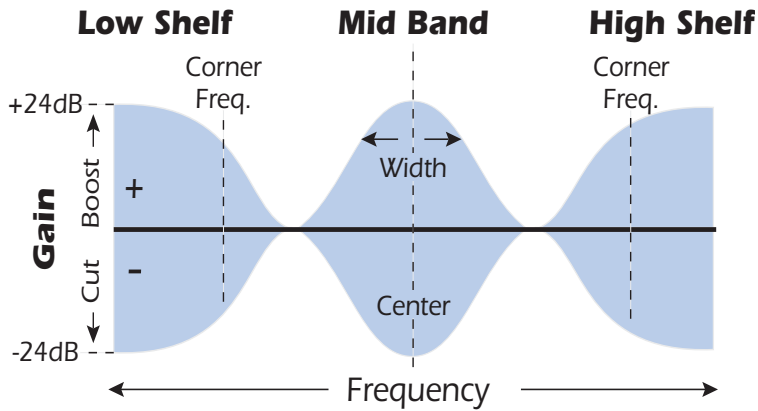
This single band shelving equalizer is useful when you just want to boost or cut a single range of frequencies at the high or low end of the spectrum. For example, if you just want to add a little more bass, there's no need to waste a 3-band EQ. Just choose low shelf, then adjust the gain and frequency. This EQ offers up to ± 15 dB 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: -15dB to +15dB

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. Now adjust the Center Frequency to “zero-in” on the frequencies you wish to boost or 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.

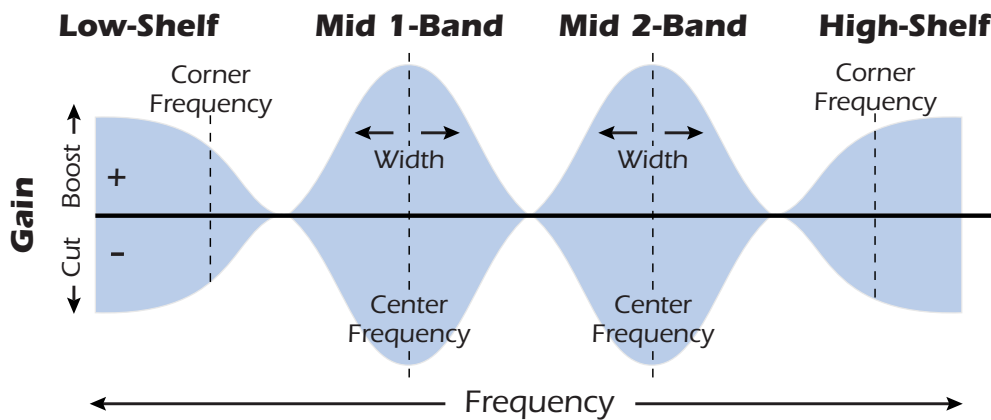
Parameter	Description
High Shelf Gain	Sets the amount of cut (-) or boost (+) of the high frequency shelf. Range: -24dB to +24dB
High Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the High Gain control. Range: 4kHz to 16kHz
Mid Gain	Sets the amount of cut (-) or boost (+) of the mid frequency band. Range: -24dB to +24dB
Mid Center Freq.	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 on setting up a parametric EQ, see [page 45](#).



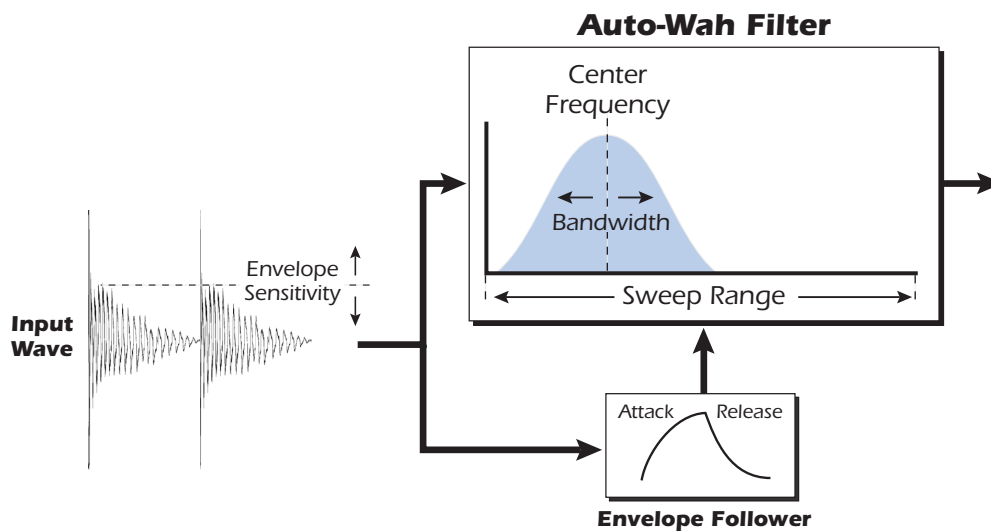
Parameter	Description
High Shelf Gain	Sets the amount of cut (-) or boost (+) of the high frequency shelf. Range: -24dB to +24dB
High Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the High Gain control. Range: 4kHz to 16kHz
Mid 2 Gain	Sets the amount of cut (-) or 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.



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: 1 Hz to 800Hz

Chorus

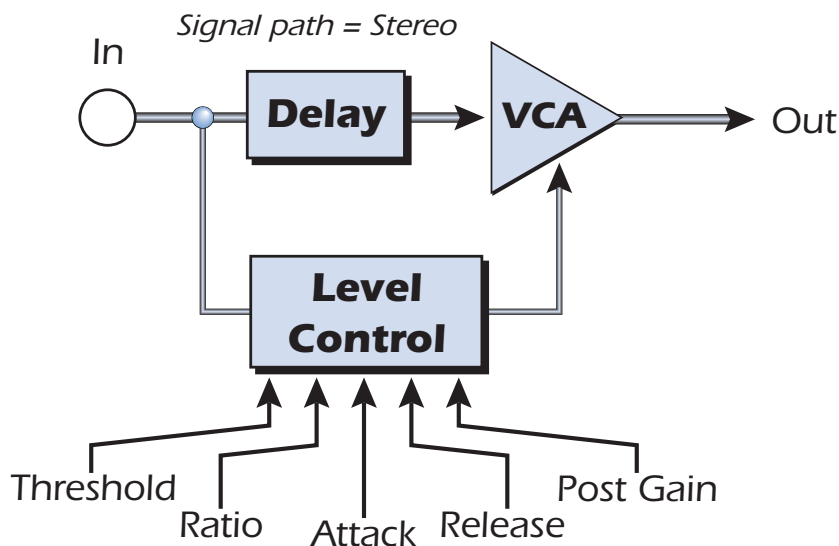
An audio delay in the range of 15-20 milliseconds is too short to be an echo, but is perceived by the ear as a distinctly separate sound. If we now vary the delay time in this range, an effect called chorus is created, which gives the illusion of multiple sound sources. A slight amount of feedback serves 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 .1 ms 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.

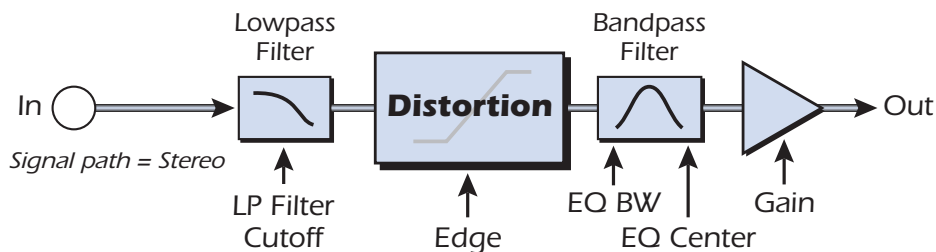
Gain Reduction Meter Shows the amount of gain reduction being applied.

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.

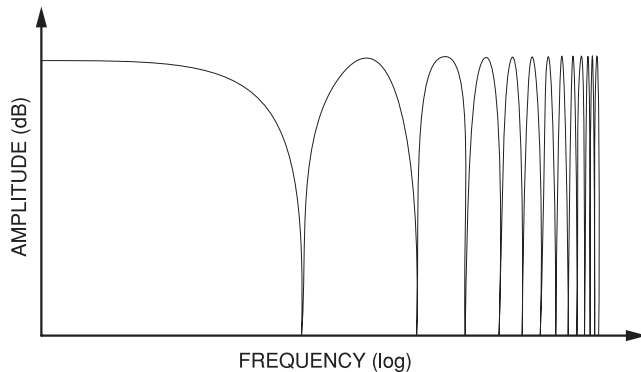
Use the Wet/Dry mix control in conjunction with the Edge control to reduce the amount of distortion, or go wild and turn everything to 11!



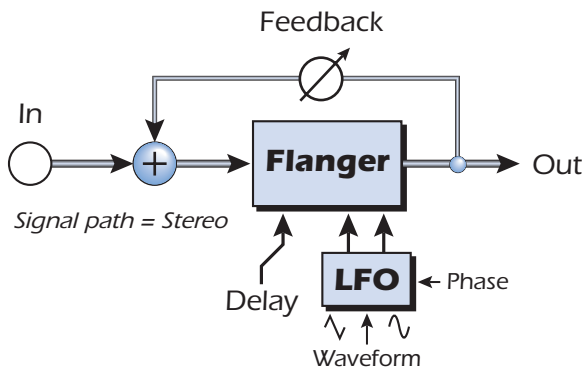
Parameter	Description
Pre EQ LP Cutoff	Controls the amount of high frequency audio admitted to the distortion. Range: 80Hz to 24kHz
Edge	Sets the amount of distortion and new harmonics generated. Range: 0-100
Gain	Sets the output volume of the effect. Range: -60dB to 0dB
Post EQ Center Freq.	Sets the 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 1/100th 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 0% 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 harmonic 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.

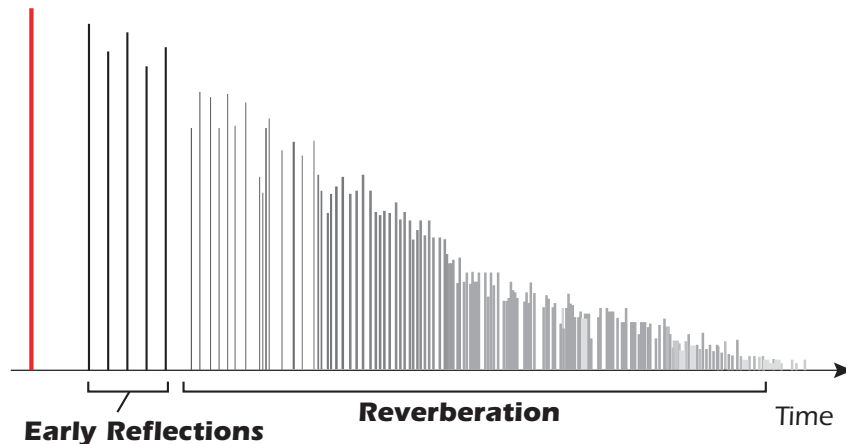
The gain reduction meter shows you how much gain reduction is being applied. Since the gain reduction meter displays how much the gain is being turned *down*, the meter moves from right to left, instead of left to right like most meters.

Post Gain	Amplifies the signal after it has been compressed to bring up the volume.
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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.



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

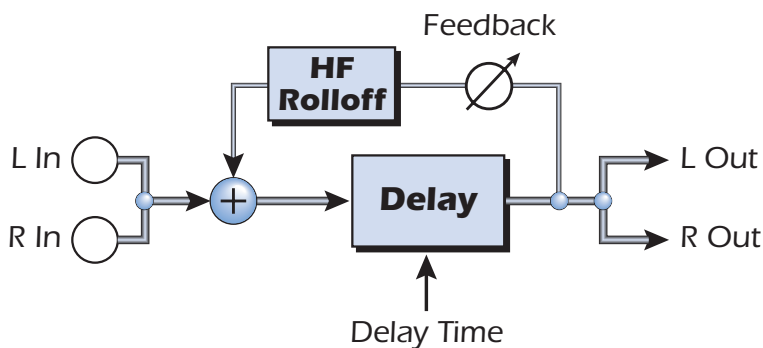
A delay line makes a copy of the incoming audio, holds it in memory, then plays it back after a predetermined time. The delay number refers to the maximum delay time that can be produced by the delay line. The six lengths, from 100 ms to 3 seconds, allow you to make the most efficient use of the effect 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. [Comb filtering: See page 51.](#)

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.

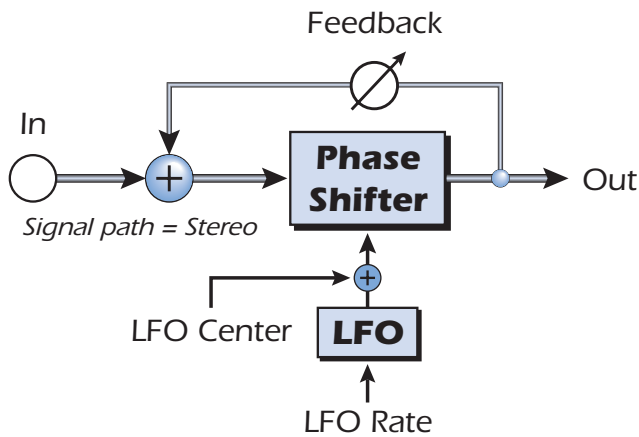


Parameter	Description
Delay Time	Sets the length of the delay in milliseconds. (.01ms. minimum increment between settings)
Mono Delay 100	Range: 1 millisecond to 100 milliseconds
Mono Delay 250	Range: 1 millisecond to 250 milliseconds
Mono Delay 500	Range: 1 millisecond to 500 milliseconds
Mono Delay 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.

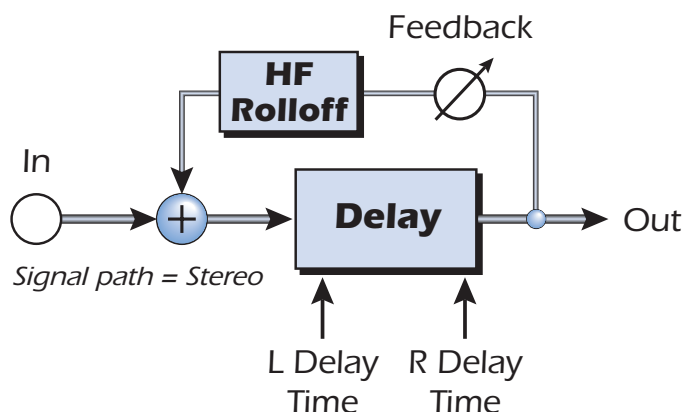
There is only one parameter on this effect. Just select the speaker you want and listen. Normally this effect should be used with the Mix control set to 100%.

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, 550, 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.

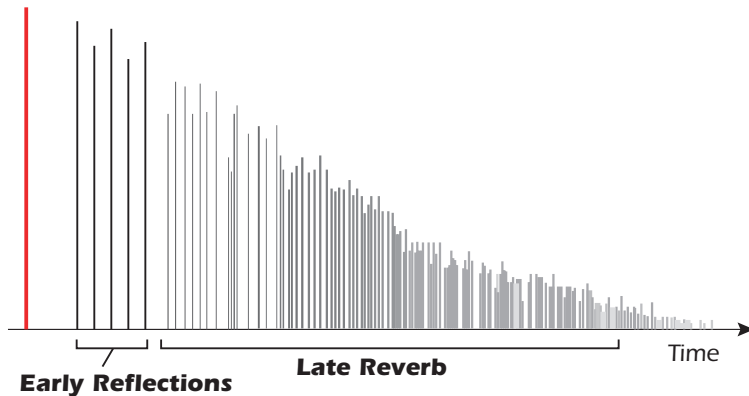


Parameter	Description
Left Delay Time	Sets the length of the delay for the left channel in milliseconds.
Right Delay Time	Sets the length of the delay for the right channel in milliseconds.
Delay Time (L & R)	<i>(.01ms. minimum increment between settings)</i>
Stereo Delay 100	Range: 1 millisecond to 100 milliseconds
Stereo Delay 250	Range: 1 millisecond to 250 milliseconds
Stereo Delay 550	Range: 1 millisecond to 550 milliseconds
Stereo Delay 750	Range: 1 millisecond to 750 milliseconds
Stereo Delay 1500	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.

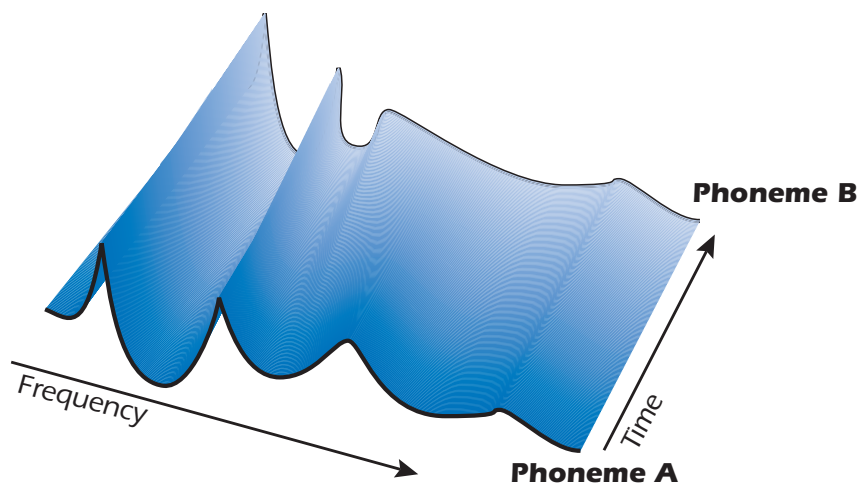
Parameter	Description
Decay Time	Sets the length of the Late Reverb. Range 1.5 to 30seconds
Early Reflections Level	Sets the volume of the initial wall reflections. Range: 0% to 100%
Early/Late Reverb Bal	Adjusts the balance between early reflections and late reverb. Range: 0% to 100%
Late Reverb Delay	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

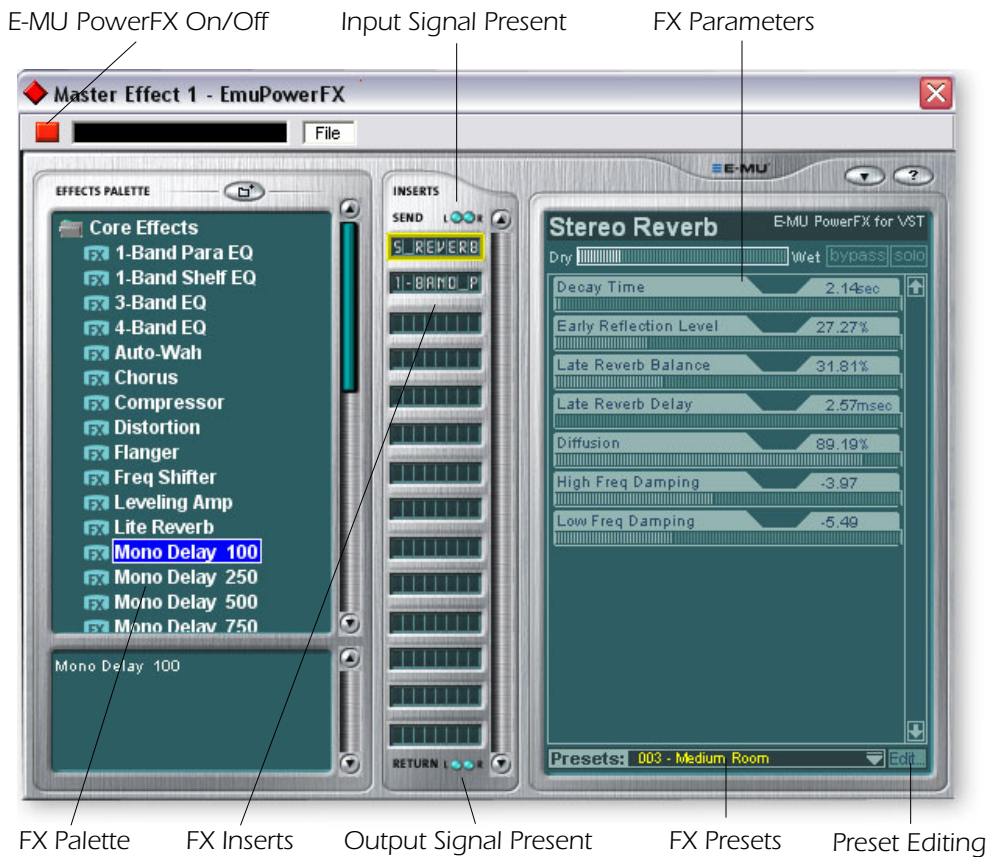
E-MU PowerFX

The hardware-accelerated effects of the E-MU Digital Audio System can also be used as VST inserts in Cubase. E-MU PowerFX allow you to use PatchMix DSP effects from within Cubase with no 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).

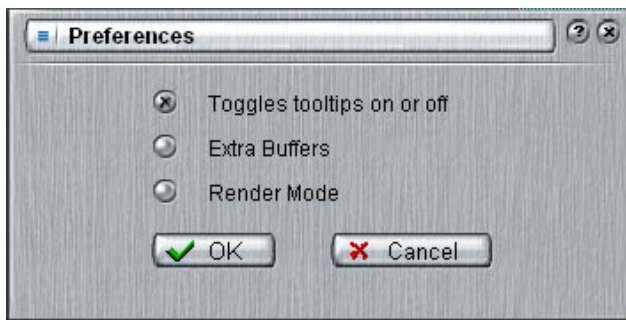
▼ E-MU PowerFX are not available at the 96kHz sample rate.

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



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: <ul style="list-style-type: none">▪ 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, SoundForge)



► **To Setup & Use E-MU PowerFX:**

Setup Cubase or Cubasis

1. Launch Cubase or Cubasis.
2. Instantiate **E-MU PowerFX** in an Insert or Aux Send location within Cubase (Look for E-MU in the VST Plug-in folder).
3. Press the Effect Edit button in Cubase to bring up the E-MU PowerFX plug-in window shown on the previous page.

Setup E-MU PowerFX

4. Press the red button (lighting it) to turn E-MU Power FX 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 highlight 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.

Add Delay Compensation (*if needed*)

If you are using Cubase VST 5.1, 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 (or other recording host) just like any other VST effect. When “Write Automation” is activated in Cubase, 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.

▼ Steinberg Cubasis does not have the control automation feature.

► To Record E-MU PowerFX parameter changes in Cubase VST

1. Add E-MU PowerFX as a Channel Insert.
2. Rewind the song and enable “Automation Write” by pressing the **WRITE** button on the VST Channel Mixer, illuminating it. (Refers to Cubase and Cubasis. 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 red “On” button is lit.
4. Press the **Play** button on the Cubase 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/Cubasis begins overwriting as soon as you change a control.
8. If you don’t like the results and want to try again, scroll to the bottom of the Arrange Window, select the “Audio Mix” track and press the **Delete** key.
9. **Note:** This erases the entire control track with any other automation. See the Cubase/Cubasis manual for specific information about automation editing.

▼ 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 new plug-in 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 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 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 PowerFX session, E-MU PowerFX plug-ins will be bypassed, since the hardware effects cannot operate at 96kHz.

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
Abelton 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 E-MU 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 SX2 or Cubase SL2. This setting will give the best results.

Tips for using Freeze Mode on Cubase SX2

- 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. See [page 62](#).
- 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 VST E-Wire

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

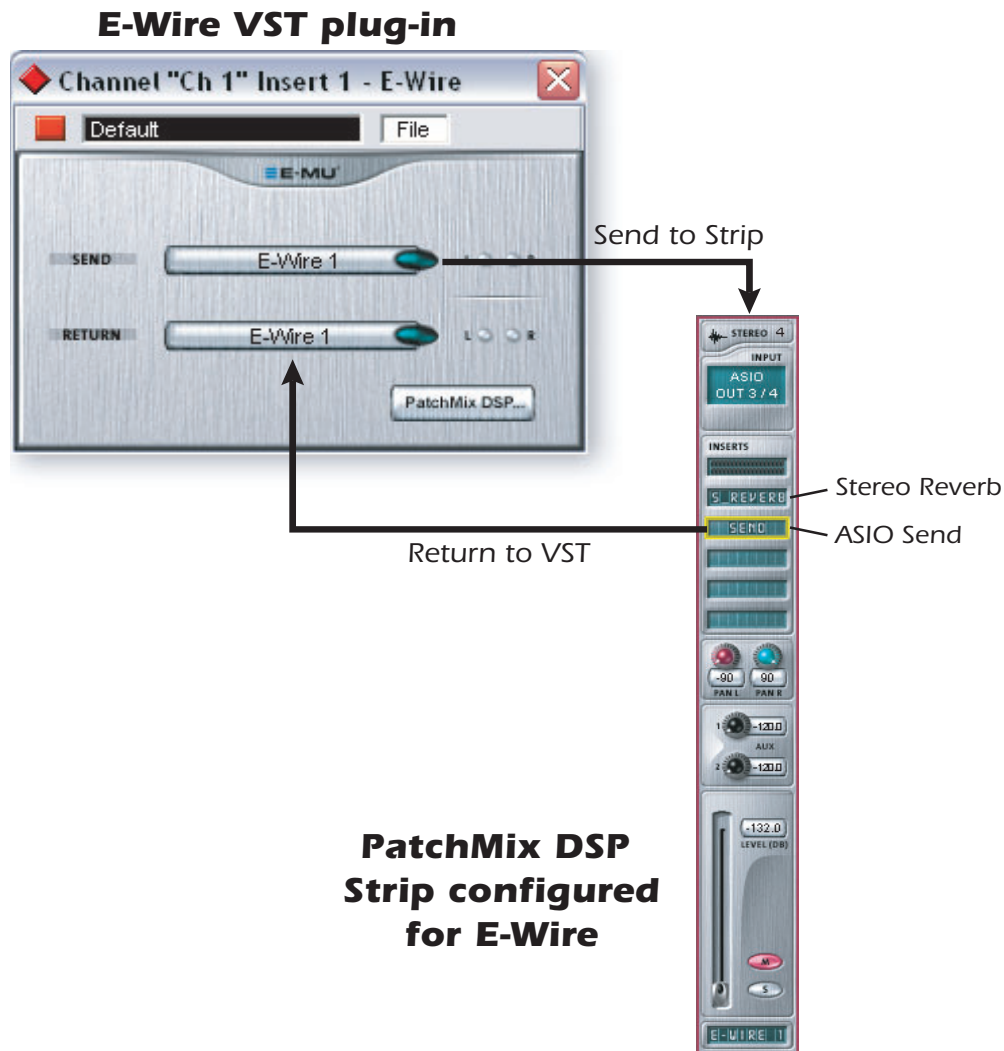
E-Wire VST incorporates smart time alignment technology that automatically compensates for system latencies and ensures proper synchronization of audio throughout the VST chain. In addition, E-Wire also allows you to insert outboard audio gear into the VST environment.

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

8. Launch Cubase.
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 red 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.



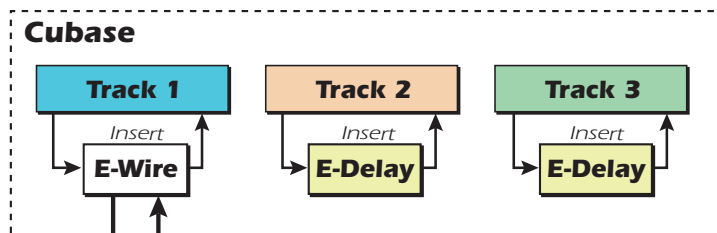
A host will support PowerFX and E-Wire's plug-in delay compensation if it supports the SetInitialDelay feature of the VST 2.0 specification.

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, unfortunately, 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 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.



**PatchMix
DSP**

E-Delay Compensator Use

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

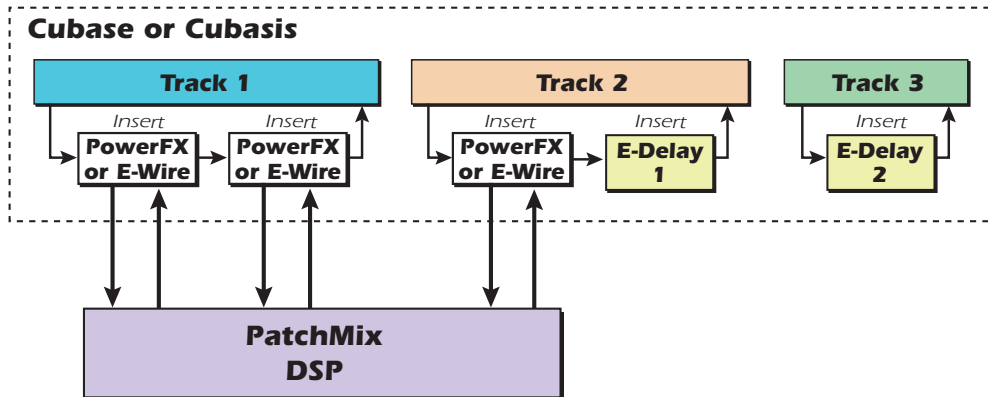
1. An E-Delay Compensator should be used when unprocessed audio tracks are played alongside tracks using a PowerFX or E-Wire plug-in.
2. Simply insert an E-Delay Compensator into each track that doesn't use a PowerFX or E-Wire send.

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 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 - Tutorials: Recording and Mixing

Getting Started with PatchMix

PatchMix = Patchbay + Mixer

It's a simple concept. PatchMix is a patchbay for your computer, where any physical input or output (analog, S/PDIF, ADAT) can be connected to any HOST input or output (ASIO, WDM, WAVE). Strips = Inputs and Insert Sends = Outputs. That's the "Patch" part of the name. Now add a simple mixer with two auxiliary sends and a bunch of effects. This lets you hear zero-latency effects while recording, a feat that no plug-in effect can match. It's a super-flexible patchbay first and a mixer second—PatchMix.

Too Much Power?

We included the PatchMix DSP mixer and powered effects because they allow you to do things that you can't do with any other sound card. However...please DO NOT be intimidated by PatchMix, because you **DON'T HAVE TO USE IT** unless you want to. The "Product Default" Session connects every input and output for you. Once this session is selected, you can forget about PatchMix and use the Digital Audio System like any other sound card without the hardware effects and flexible routing features. PatchMix was never meant to replace the mixer of your recording application. You may not need this additional power right away. Use PatchMix in the background if you prefer, and come back to it whenever you're ready.

ASIO

ASIO stands for Audio Stream In/Out and is a standard communication protocol between audio software and a piece of audio hardware. The low latency made possible by ASIO, for example, allows you to sing into a microphone and hear your voice going through software and hardware effects with virtually no delay in the signal. The effects of a low-latency ASIO driver make it ideal for use with Virtual Studio Technology (VST) instruments, plug-ins, and all other software-based synthesizers.

IMPORTANT -> Activating ASIO

It is important to understand that ASIO inputs do not appear in Cubasis or other host applications until they are **CREATED** in PatchMix DSP. This allows you to dynamically assign the 32 available ASIO channels according to your specific needs.

► For Example:

1. Create a new strip from a physical source such as PCI Card S/PDIF L/R.
2. Next, right-click on the new strip's insert chain and select the **Insert Send** option.
3. Choose an available host ASIO input.
4. This S/PDIF L/R input should now be available to record in your ASIO application.

❖ If the Insert Send comes *after* an effect, the effect will be recorded.
[See Recording & Monitoring with Effects.](#)

ASIO Channels are STEREO!

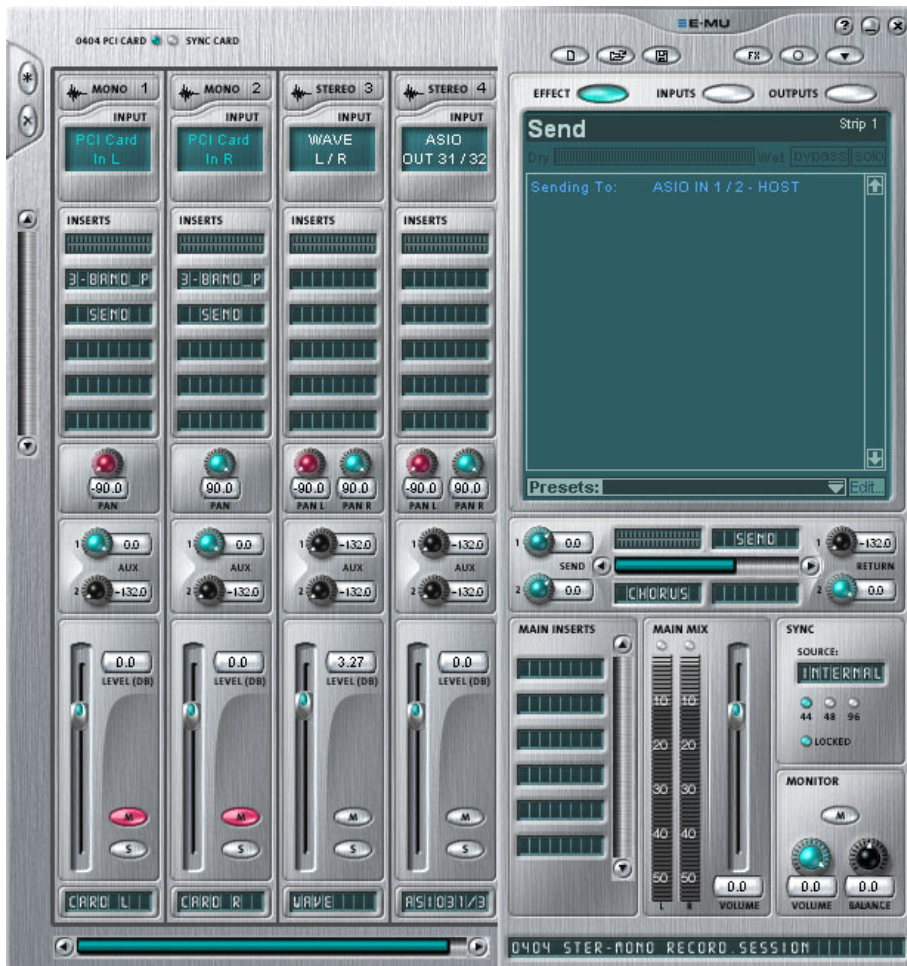
ASIO channels are stereo, period. Even when you are recording a mono source such as a microphone, the ASIO stream is stereo (with the mono signal on both channels). To record a mono input, you simply set up the Cubasis track as a mono track and use either the left or right ASIO channel from PatchMix as the input for the track.

► To Set Up a Stereo Recording Session:

It's best to combine the PCI Card Left and Right inputs into an ASIO pair using one of the Aux buses. Because the panning network comes *BEFORE* the Aux Sends you can pan the two signals left and right then combine them into a ASIO stereo pair using the Aux 1 summing bus. We have provided a special PatchMix DSP Session Template for this purpose entitled, **44.1 Mono-Stereo** session. You can simply open this Template and use it for either mono or stereo recordings. Here's how it works for Stereo:

1. Create **Pre-Fader PatchMix** strips for the **PCI Card In L** and **PCI Card In R**.
2. Create a **Host ASIO Output Source PatchMix** strip. (*ASIO OUT 31/32 is used in the factory template.*) This channel will used to monitor the output from Cubasis
3. Pan **PCI Card In L** full **Left** and **PCI Card In R** full **Right**.
4. Set the **Aux Send 1 Amount** to **0.0** on **PCI Card In L** and **R** channel strips.
5. Set the **Aux 1 Bus Send Amount** to **0.0** in the **Aux 1 Bus**. (*Sending to ASIO 5/6 HOST for example.*)
6. Place an **Insert Send** (Output to ASIO) into the **Aux 1 Bus**.
7. Set the **Aux 1 Bus Return Amount** to **-132.0** in the **Aux 1 Bus**. (So it won't feed the main bus.)
8. **Mute the Faders** on the two strips because you will monitor through Cubasis.

Mono-Stereo Session Template



► **To Set Up a Mono Recording Session:**

To record each input as a mono track, you just place an ASIO Send into each channel's Insert Section. Because ASIO is stereo, the mono inputs will be duplicated on both of the ASIO pair. The PatchMix DSP Session Template **44.1 Mono-Stereo Session** is already set up for you. Open this Template and use it for either mono or stereo recordings. Here's how the template works for Mono:

1. Create **Pre-Fader** PatchMix strips for the **PCI Card In L** and **PCI Card In R**.
2. Place an **Insert Send** (Output to ASIO) into the left and right strip.
(*ASIO 1/2 - HOST, & ASIO 3/4 - HOST are used in the factory template.*)
3. Panning doesn't matter in this example since the panning network comes *after* the Insert section.
4. **Mute the Faders** on the two strips because you will monitor the inputs through Cubasis.

Basic Recording

The following step by step tutorials are designed to get you recording as quickly as possible. After you have finished the tutorials we encourage you to read the Cubasis pdf manual in order to learn about the many features of this powerful program. We have provided special PatchMix DSP Sessions, as well as a Cubasis Recording Template Song to go along with the following tutorials.


Recording into Cubasis

Follow these instructions if you have an 0404 system and Cubasis is installed on your computer. In this tutorial, we'll assume you're using just a single input or pair of inputs. For more advanced recording, refer to the Cubasis manual.


To use PatchMix with Cubasis, you must have 4 basic components in PatchMix:

- a) a physical input strip for your input source.
- b) a send from this strip to an ASIO input so that Cubasis can receive your input.
- c) an ASIO output strip for Cubasis to output its audio through.
- d) an output assigned to any physical output.

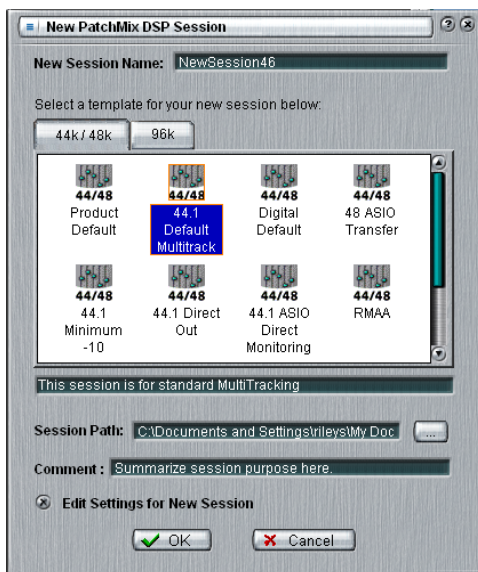
Open the PatchMix DSP Mixer

1. Open the PatchMix DSP mixer by clicking on the  icon on the Windows Taskbar.

To Load the Default Multitrack Session

2. Click the "New Session" button,  which is the upper left button above the TV screen. The following screen appears:

❖ Because unused strips waste DSP resources needlessly complicate the mixer display, you may want to delete the strips you aren't using.



3. From the 44k/48k tab, select **44.1 Mono-Stereo** session. This session is all set up to record in either mono or stereo. It also has a WAVE strip and one pair of ASIO outputs (31 & 32) used for monitoring the main output of Cubasis.
4. The **Session Settings** screen appears showing 44.1kHz. Click **OK**.
5. If you wish to record in stereo, connect a preamplified instrument or microphones to inputs L/R. To record in mono, connect to either the left or the right input.

Open Cubasis

6. **Open Cubasis** by clicking on the desktop shortcut.
7. On Cubasis, select **File, Open**.
8. If you want to record in **Mono** open the Cubasis Arrangement **(0404) Mono-Stereo Template** which is located here: (My Computer\Local Disk-C\Program Files\Creative Professional\PatchMix DSP\VST\Recording Templates).

Select the Audio Files Folder

Choose the location where your audio files will be stored. It's usually a good idea to do this step first as if you were, "Putting the tape reels on your recorder."

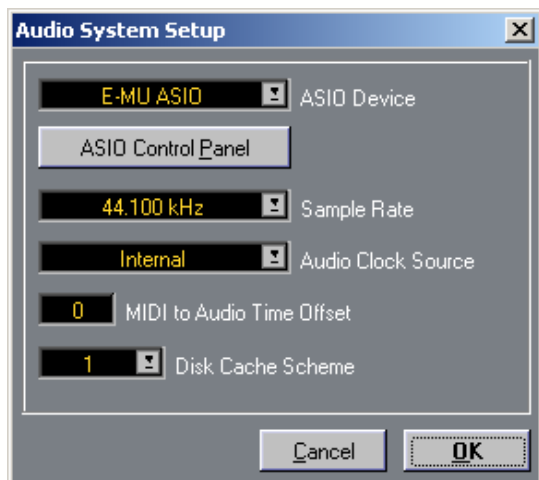
9. Go to **Options, Audio Setup, Audio Files Folder**.
10. Choose any location that you will be able to find later. A fast, "non-boot" drive is usually preferable for recording audio files.

Setup Cubasis Audio & ASIO Settings

11. Set the Audio System Settings and ASIO Settings as shown below.

Cubasis Audio System Setup Settings

(Options, Audio Setup, Audio System Setup...)



The dialog box above shows the correct settings for 44.1 kHz.

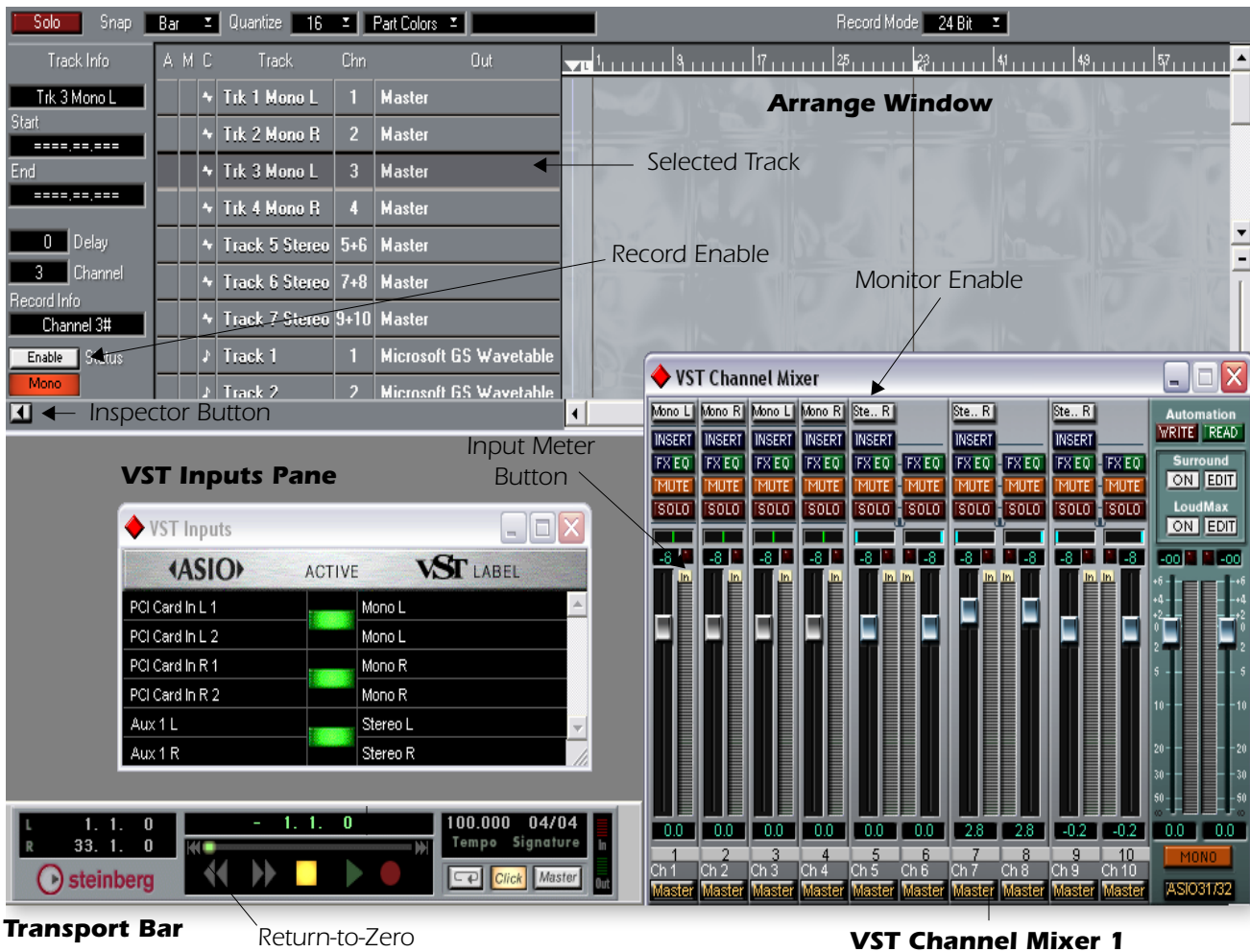
E-MU ASIO Control Panel Settings

Click the ASIO Control Panel button to set the ASIO Buffer Latency.



12. ASIO Buffer Latency should be set to 5 milliseconds or less, depending on the speed of your computer's CPU. This control adjusts the amount of delay incurred while transferring audio between the Digital Audio System hardware and Cubasis.
13. Turn **Enable Monitor** on. This is located under **Options, Audio Setup**.

The Cubasis Demo Song Screen



Set up your Inputs to Record

14. Open the **VST Inputs** from the **Panels** menu. Notice that all the Sends from PatchMix DSP are listed here. The green buttons are used to turn the Cubasis inputs on and off. Turn all the inputs on if they aren't already on.
15. Click the **Input Meter** buttons on all the VST Channel Mixer strips, illuminating them. This turns the meters on so you can verify that you're getting an input signal.
16. Make sure your input channels are getting into Cubasis by examining the **VST Channel Mixer 1**. You should see the meters move when you play your instrument. If not, first check the input meter on PatchMix DSP. If you still don't see meter activity, refer to the "[ASIO Recording Checklist - Cubasis](#)" on page 77. To change the input on a Cubasis input mixer strip, hold **Ctrl** and click the top button in the strip.
17. Click the **Monitor Enable** button to hear the return from Cubasis. You should see activity on the Cubasis Master strip when you play your instrument.
18. Turn **Record Enable** On for each track you want to record. See the screen above

❖ **Hidden Feature:**
Open the **Track Info** section by clicking the **Inspector** button in Cubasis. See above.

19. Select the **Track** that you want to record by clicking on it in the arrange window. The rack darkens to indicate that it is selected. See the “Cubasis Demo Song Screen” above.

Record a Track

20. Press **Record** on the Cubasis Transport control and start playing. There will be a two bar count-in before recording starts.
21. Press **Stop** (or hit the space bar) when you’re finished recording the first track.
22. Press the **Return-to-Zero** button on the Cubasis Transport (or tap the space bar twice).
23. Press **Play** on the Cubasis Transport to play back your track.

Record Another Track

24. **Return to Zero** by tapping the space bar twice.
25. **Select the next track** to record by clicking the appropriate track in the Arrange window. Remember to choose the appropriate side (L or R) if recording in mono or choose a stereo track if recording in stereo.
26. Press **Record** on the Cubasis transport and start playing. You’ll hear the first track as well as what you’re playing.
27. Press **Stop** (or hit the space bar) when you’re finished recording.

❖ You can turn the metronome off by pressing the **Click** button on the Transport control.

Double-click the ‘Click’ button to edit the click and countoff settings.

❖ **Cubasis Tip:** If you’re recording several tracks using the same inputs, you can simply drag a recorded Part (audio chunk) down to the next track then start recording again without changing input settings.

ASIO Recording Checklist - Cubasis

- An **ASIO Send** is inserted into a PatchMix DSP Mixer strip.
- E-MU ASIO** is selected (Options, Audio Setup, Audio System Setup)
- ASIO Buffer Latency** is set to 5 ms or lower (depending on the speed of your computer). (Options, Audio Setup, Audio System Setup, ASIO Control Panel)
- Play in Background** enabled. (Options menu)
- Enable Monitor** is selected. (Options menu, Audio Setup)
- Record Enable** is turned on for the track you want to record. (Use Inspector button, then click on Enable, illuminating the button.)
- VST Inputs** are selected. (Panels menu, VST Inputs)
- Bring up the **VST Channel Mixer**. Turn Input Meter button on.
- Master Mixer output** assigned to ASIO 31/32.
- An **ASIO Strip** on ASIO 31/32 exists in the PatchMix DSP Mixer.
- Monitor** button is selected on the channel you are recording. See the diagram on [page 76](#)

Using PatchMix DSP with Sonar 3

The following instructions show you how to set up PatchMix DSP & Sonar 3 for recording through an analog input.

In this example, we will use the **PCI Card In L** as the input source. If you are using a guitar or other low-level instrument source, it must be properly pre-amplified in order to achieve a usable signal level.

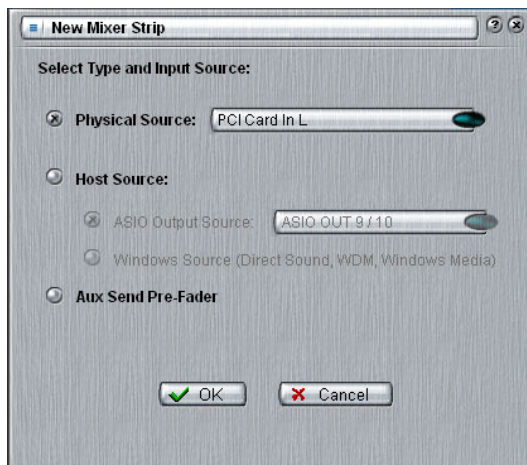
1. Plug in your input source.
2. **Launch PatchMix DSP** by clicking the **E-MU** icon in your system tray next to your clock. Press the **New Session** button, then select the "44.1 blank" session file.



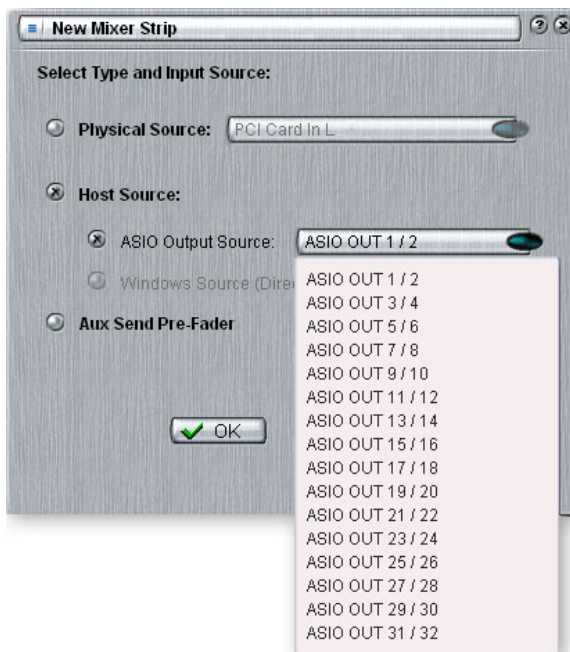
3. To use PatchMix properly with Sonar 3 (and other supported ASIO software), you must have 4 basic things in PatchMix:
 - a) a physical input strip for your input source.
 - b) a send from this strip to an ASIO input so that Sonar can receive your input.
 - c) an ASIO output strip for Sonar to output its audio through PatchMix DSP.
 - d) a PatchMix DSP output assigned to any physical output (to your monitors).
4. Insert the **PCI Card In L** as a physical input strip by pressing the **Create Strip** button in the upper left.



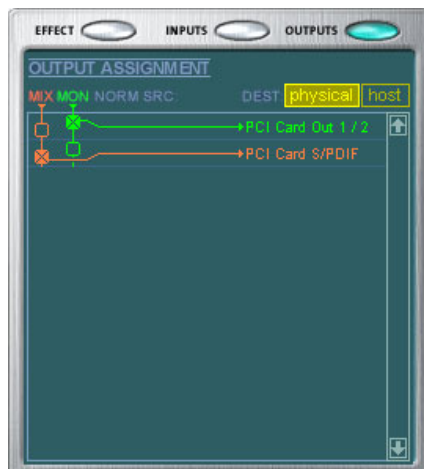
5. Choose **Physical Source**, press the button for the drop-down box and choose **PCI Card In L**, then press **OK**.



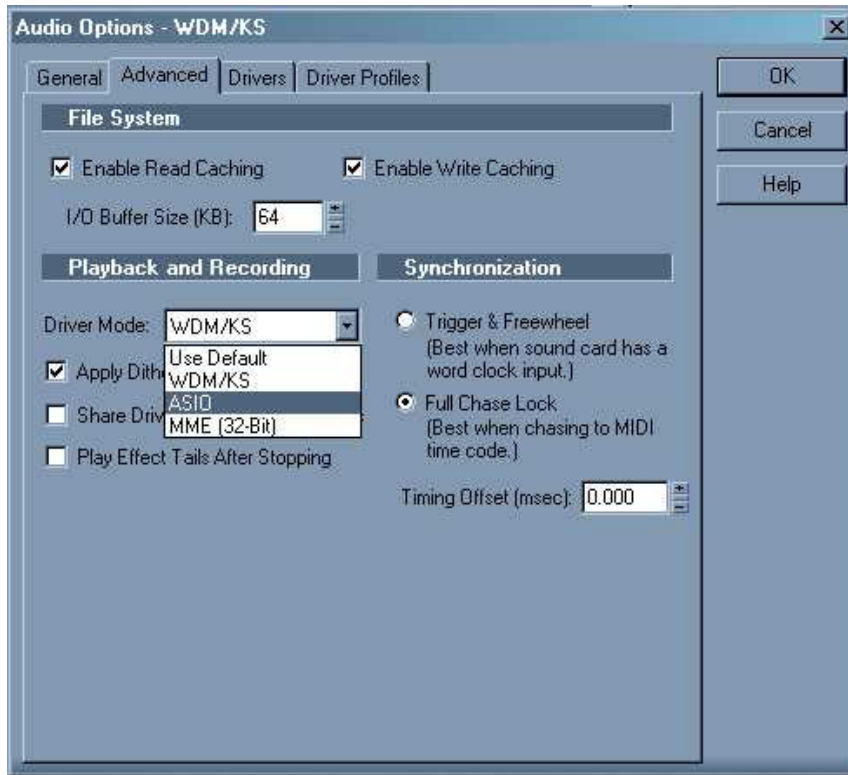
6. Turn your **preamp knob clockwise** and check your instrument. You should begin to see signal in the Main Mix peak meters. **Right-click** the first blank insert slot on the PCI Card In L strip and choose **Insert Peak Meter**. You can now see your input level on the strip itself.
7. **Right-click** the next blank insert slot on the PCI Card In L strip (below your peak meter) and choose **Insert Send** (Output to ASIO/WAVE or Physical Out). Press the drop-down box and choose **HOST ASIO IN 1 / 2** then press **OK**. The word **SEND** will now appear as an insert on this strip.
8. Insert an ASIO output strip by pressing the **Create Strip** button. Choose **Host Source**, then **ASIO Output Source**, then press the drop-down box button and choose **ASIO OUT 1 / 2**, then press **OK**.



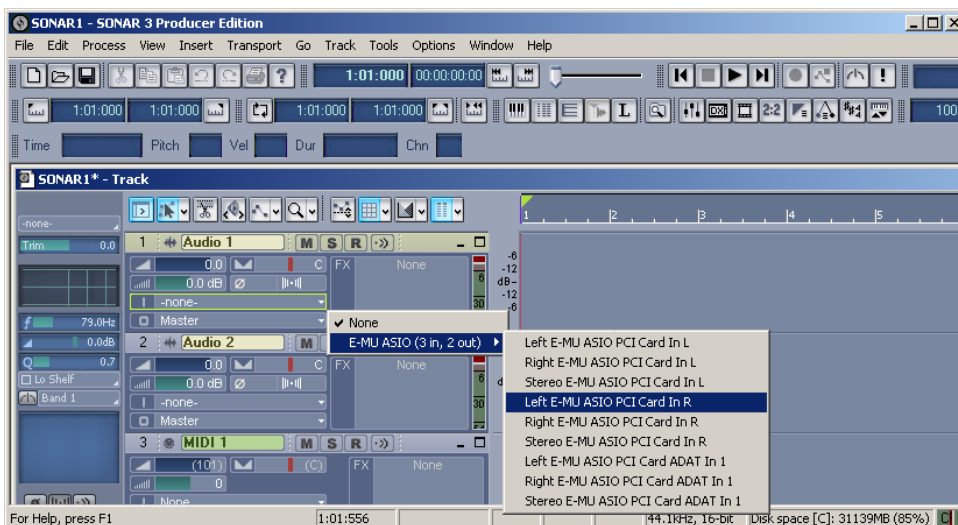
9. Now we can assign your physical monitor output(s). Press the **OUTPUTS** button above the TV screen, then the yellow **physical** button. Determine which output your monitors are plugged into, then put a check in the corresponding green **MON** box. For instance, if your monitors are plugged into the two analog outputs, choose **PCI Card Out 1/2**.



10. **Minimize PatchMix and launch Sonar 3.** Set your driver mode to ASIO by pressing the **Options > Audio > Advanced** tab. Press the drop-down box for Driver Mode: and choose **ASIO**, then press **OK**. You will have to restart Sonar for the changes to take effect. After re-launch, go to **Options > Audio > General** tab. Set the Playback Timing Master: to **E-MU ASIO ASIO OUT 1 / 2** and the Record Timing Master: to **E-MU PCI Card In L**, then press **OK**.



Start a new project and select the **Audio 1** track. Set the input source as E-MU ASIO (1 in, 1 out) > Stereo E-MU PCI Card In L. Press the track's **R** button to arm the track for recording and ensure that you are getting signal in Sonar's peak meter. You can now Press the record button and begin.

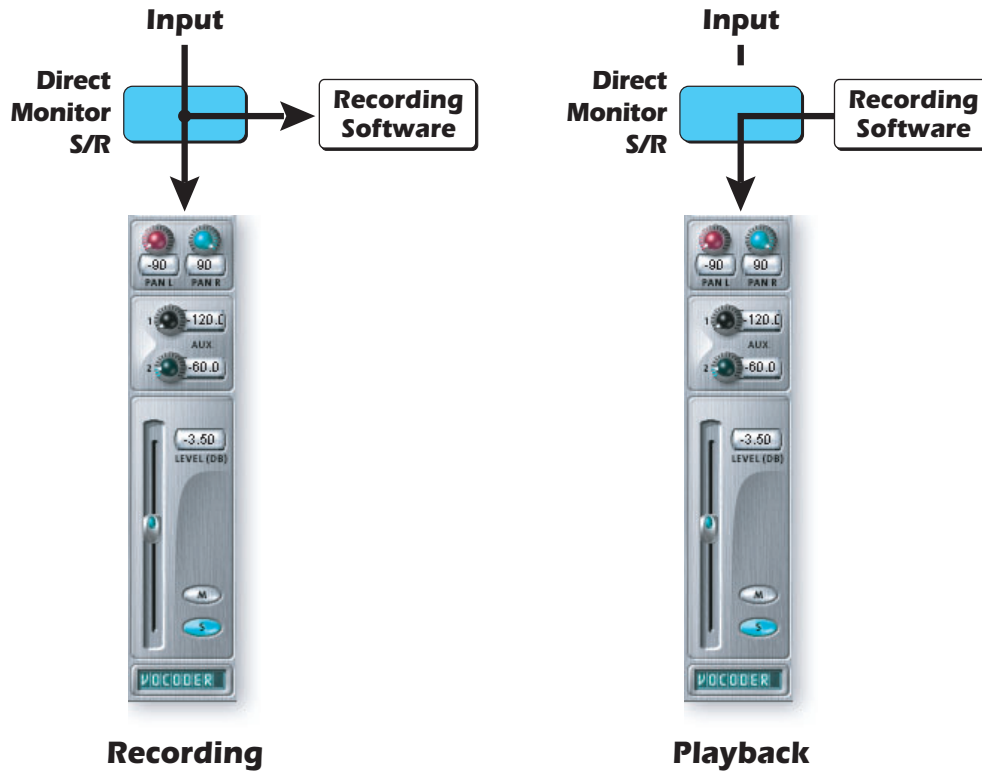


Direct Monitoring

Direct Monitor Sends create a different type of ASIO path into your host application. The direct monitor send *breaks* the signal at the insert point and sends it out to the selected ASIO Host Input destination (such as Cubase). A return source signal is also selected which *returns* the signal to the channel strip from an ASIO Host Output.

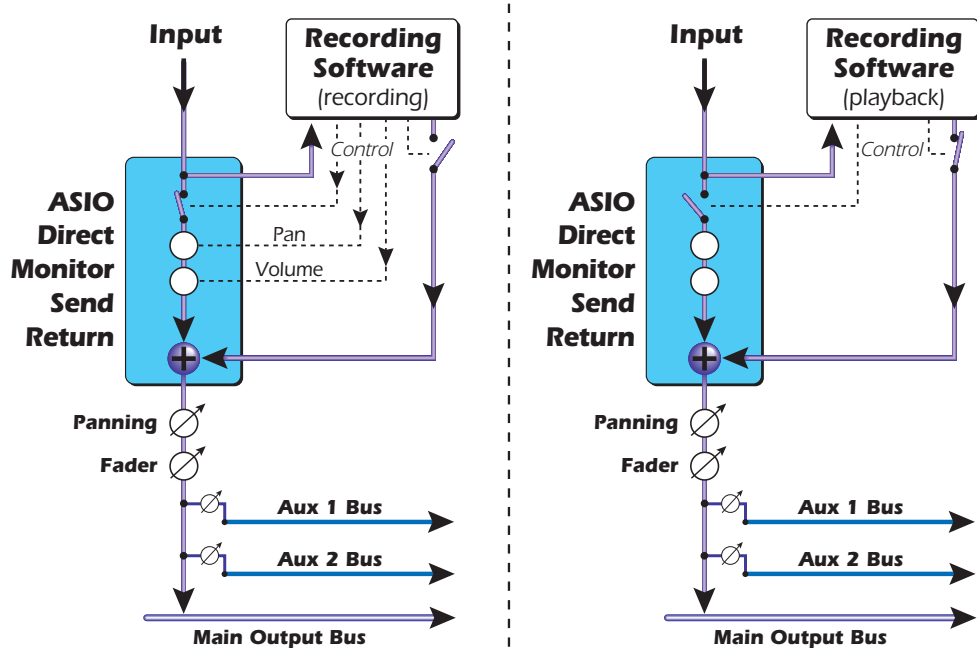
While recording, the direct monitor send/return routes the signal to the recording application, but monitors directly from the input to eliminate latency. During playback, the recording application automatically switches the direct monitor send/return to monitor the recorded track.

■ **Note:** Cubasis does not support ASIO Direct Monitoring. This section may be useful if you upgrade to Cubase SL or another recording application which supports Direct Monitoring.



The direct monitor send/return also allows the recording application to control volume and pan (this occurs before the Patchmix DSP volume and pan controls). Normally when using direct monitor recording you'll want to control the volume and pan from the recording application. In this case, set stereo pan controls hard left and right, mono pan controls to center, and the fader to 0dB in the PatchMix DSP strip.

Another view of Direct Monitoring



Pan and Volume are also controlled by your recording application via the Direct Monitor Send. These controls come before the volume and pan controls in the channel strip.

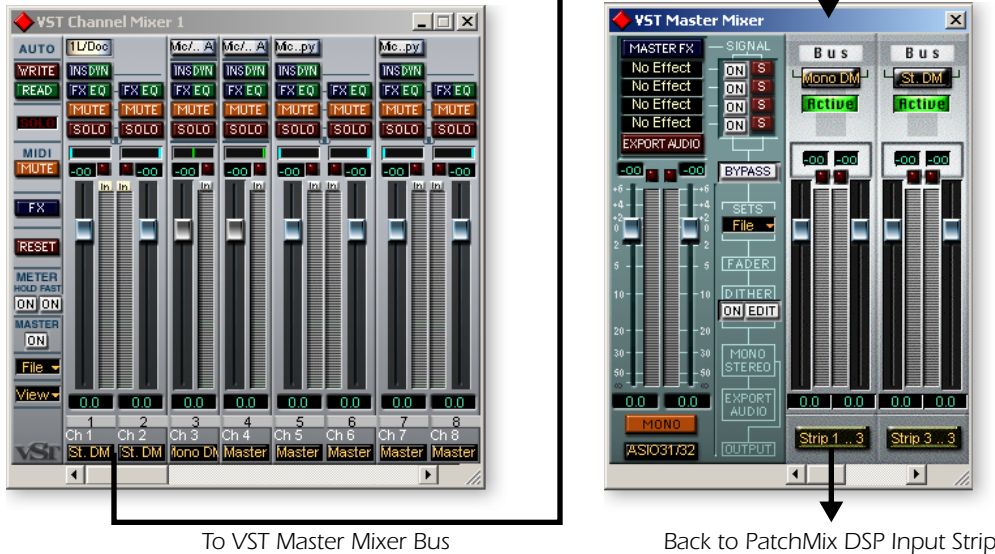
► Add a Direct Monitor Insert to an Input Strip:

1. Create a new strip from a physical source such as PCI Card In L.
2. Next, right-click on the insert chain and select the **Insert ASIO Direct Monitor** option.
3. Choose an available **ASIO Input and Output** pair. Normally, you should use the same ASIO pair for the input and output.
4. The **PCI Card In L** input should now be available to record in your ASIO application.
5. In Cubase, **ASIO Direct Monitor** needs to be **On** (checked). (Options, Audio Setup, System.)
6. In Cubase, the VST Channel Mixer input channel must be routed to a Master Mixer Bus and the output of this Bus must be routed to the PatchMix input strip containing the ASIO Direct Monitor send. This routing is shown in the diagram on the following page.

❖ If the Direct Monitor Send/Return comes *after* an effect, the effect will be recorded. [See Recording & Monitoring with Effects.](#)

Direct Monitor Routing In Cubasis

(back to PatchMix DSP)

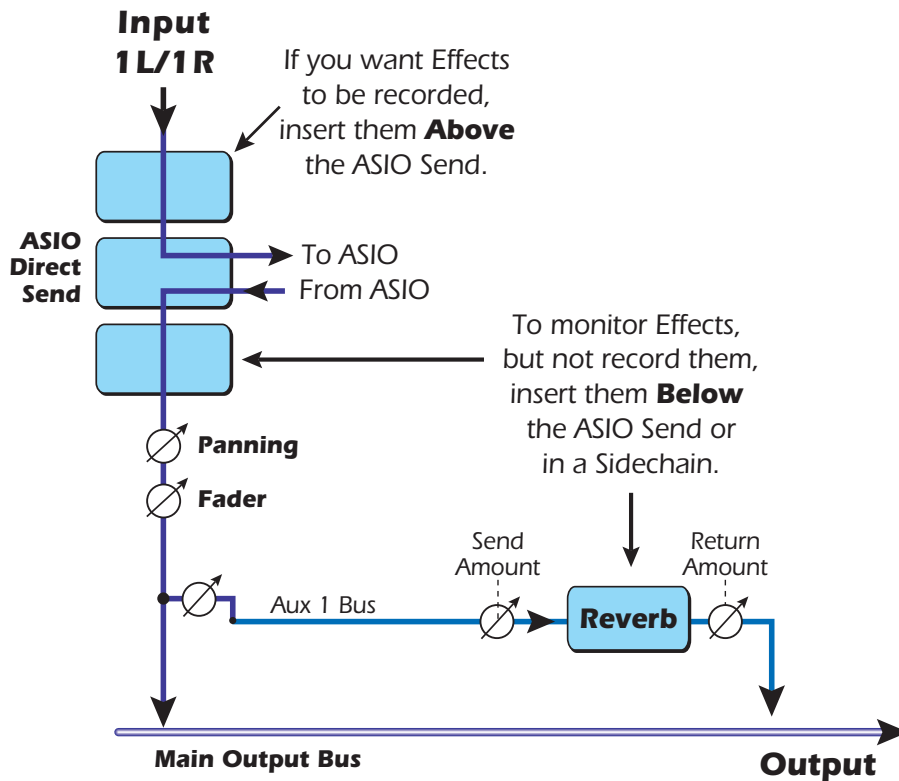


To VST Master Mixer Bus

Back to PatchMix DSP Input Strip

Recording & Monitoring with Effects

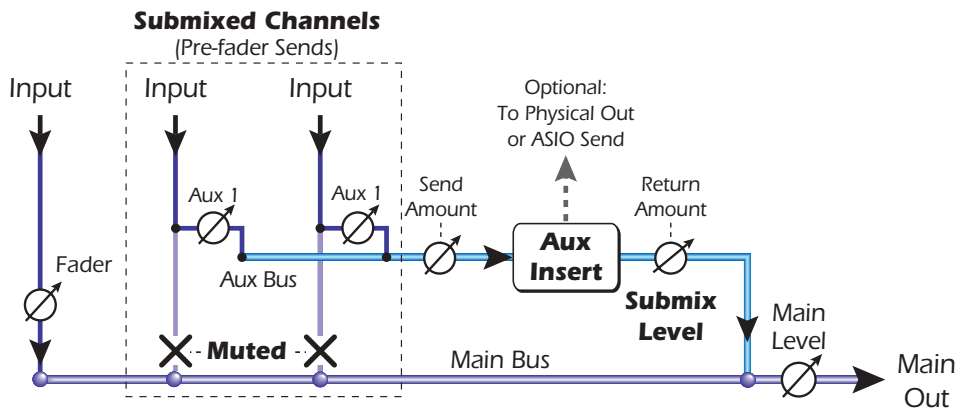
You can record a dry signal while monitoring yourself with effects! PatchMix DSP allows you to record your tracks without effects (dry) and monitor with effects enabled. This allows the musician to hear the performance with effects, but gives you the flexibility to add or modify the effects later during mixdown. The basic session setup for this is shown below.



Connection Setups

Creating a Submix

Submixes are normally used to control a group of inputs from a single fader or to process several inputs with the same effect. In most cases you will create submixes in your recording application such as Cubasis. You can also create a submix in patchmix DSP using either of the two Aux buses.

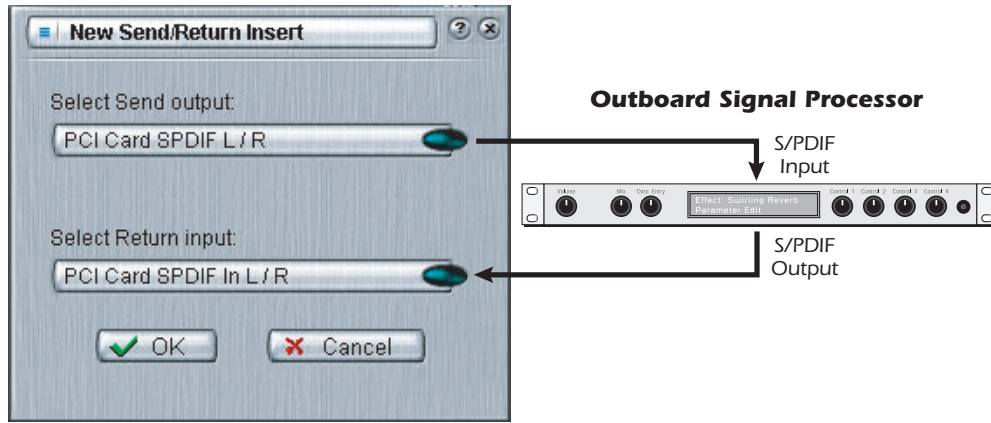


When using one of the Aux Send buses to create a submix, the Send Amounts in the strip take the role of the Channel Fader. The Aux Return Amount controls the overall level of the submix going back to the Main Mix.

Control the amount of signal from each strip using the Aux Send control and use the Aux input or output knob to control the overall volume of the submix. The submix output can be mixed back into the main mix (using the Aux Return control) or sent anywhere else by placing a physical or ASIO send into the Aux Insert chain.

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. You can insert a level meter after the Send/Return to make sure you're getting a strong signal back from the external device.
4. 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 16.](#)

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 or Cubasis.

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.

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 (Cubasis, etc.). MTC is also available at the MIDI jack on the back of the Sync Card.

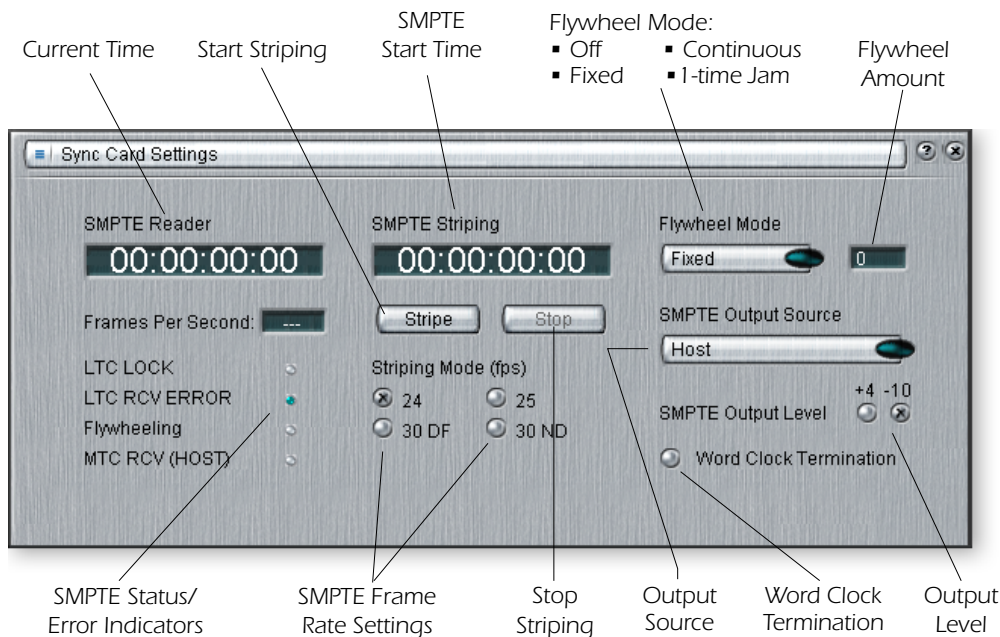
▼ **Warning:** SMPTE and MTC do not provide sample sync for digital I/O. You must use Word Clock, or S/PDIF sync.

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 flywheel 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 "Word Clock In/Out" .
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 outputting 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](#).

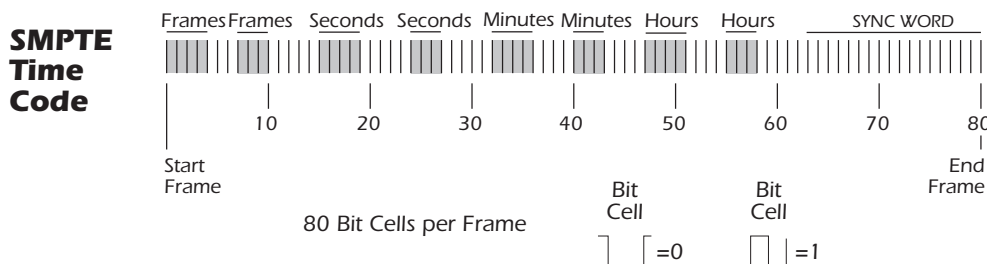
SMPTE Background

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.

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, 30 non-drop, 30 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.

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.

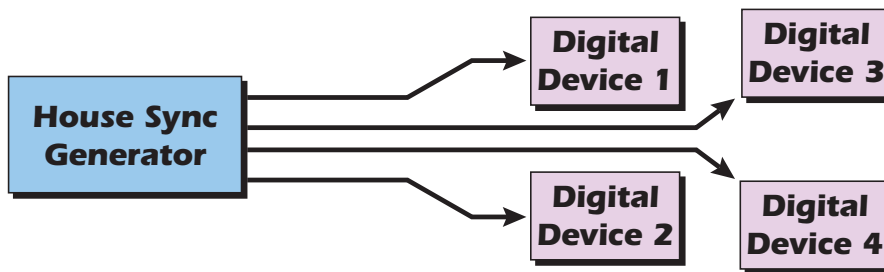
The E-MU 0404 PCI card can be externally clocked from the S/PDIF input (either optical or electrical) or from the Sync Daughter card (if installed). **In a digital studio, all digital devices in the system should run off the same master Word Clock.**

❖ Word clock and S/PDIF synchronize at the sample rate and are used to transfer digital data between machines.

► 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 or S/PDIF input.
2. Open the **Session Settings** dialog box.
3. Under the **System** tab, select **External Source**, then select either **Word Clock** 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.



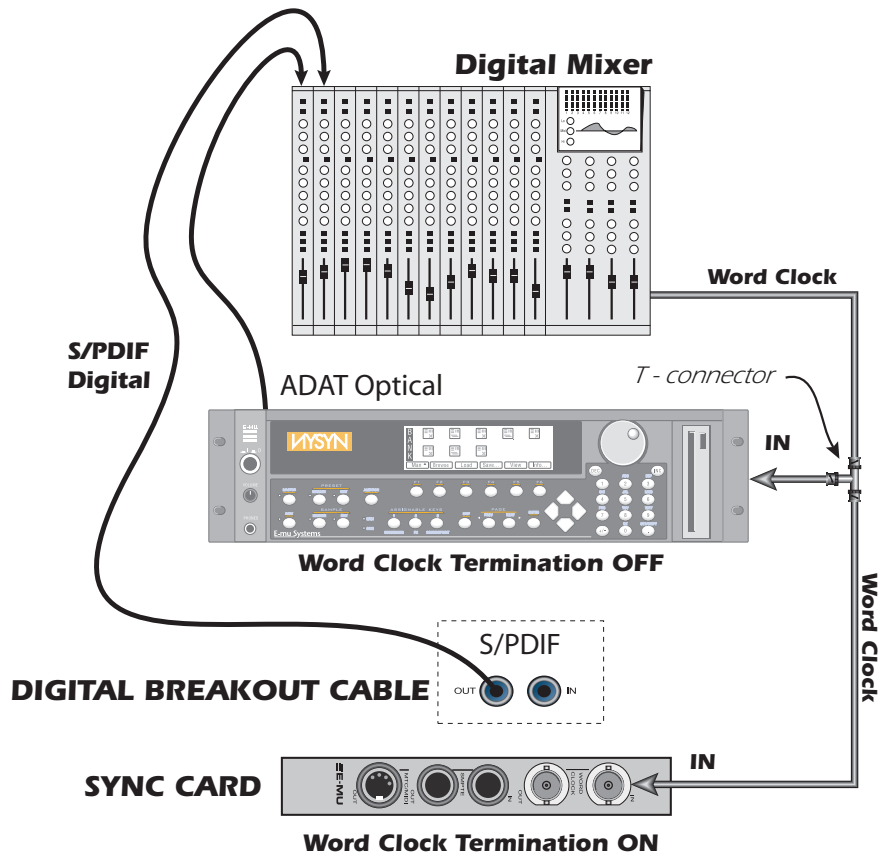
A master word clock generator is preferable for larger digital setups.

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.

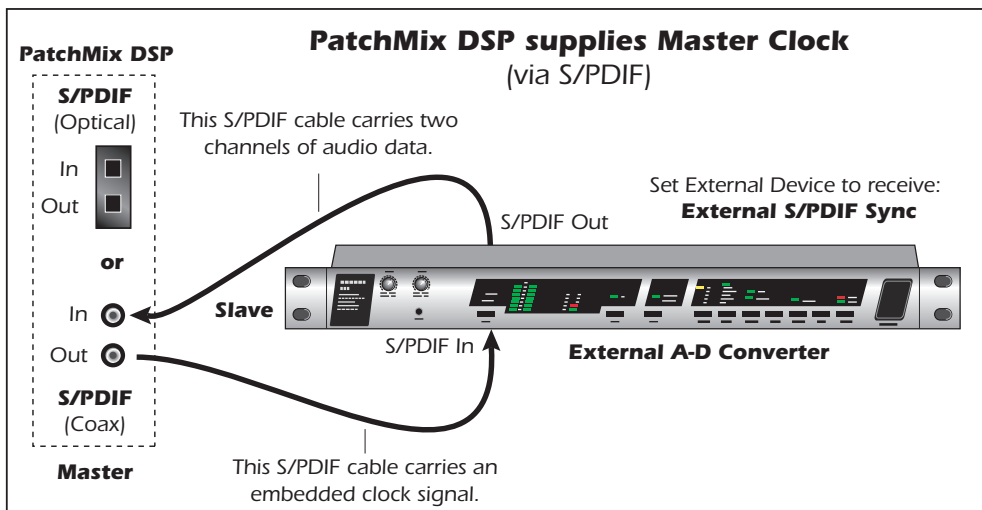
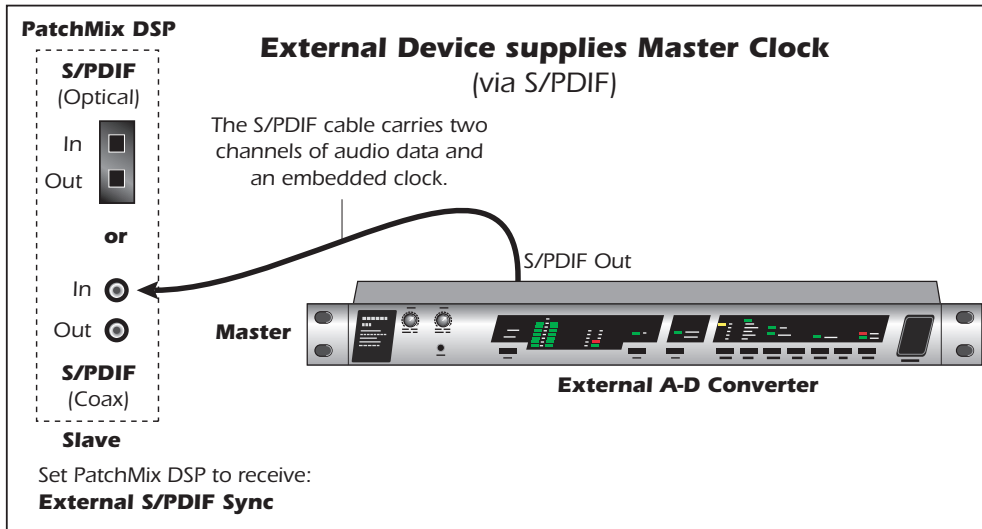
Getting in Sync

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 is probably the most common digital audio format. S/PDIF carries 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 converter to the E-MU Digital Audio System using the S/PDIF connection.

In the first example, the external A/D converter is the master clock for the system. Only one S/PDIF cable is needed (either optical or coaxial) 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.

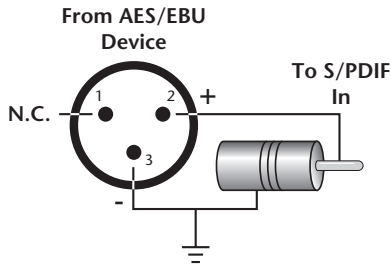


In the second example a second S/PDIF cable is used to supply "embedded word clock". The external device MUST be set to receive external clock via S/PDIF or the units will not be synchronized. The E-MU Digital Audio System is the Master and the external A/D is the Slave.

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 0404 PCI card. This cable may also work to connect S/PDIF out from the 0404 digital breakout cable to the AES/EBU input of other digital equipment.



Digital Cables

Don't cheap out! Use high quality optical fiber and low-capacitance electrical cables 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.

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

GENERAL

Sample Rates	44.1 kHz, 48 kHz, 96 kHz derived from internal crystals <i>(no sample rate conversion is performed)</i> Externally supplied clock from S/PDIF. (or word clock with optional Sync Card)
Bit Depth	16-bit or 24-bit <i>(depending on the setting of your recording or audio application)</i>
Hardware DSP	100MIPs custom audio DSP. DSP - 32-bit integer math with a 67-bit accumulator PCI Bus-Mastering DMA subsystem reduces CPU usage. Zero-latency direct hardware monitoring with effects
Converters & OpAmps	ADC - PCM1804 (TI/Burr-Brown) DAC - AK4395 (AKM) OpAmp - NJM2068M (JRC)

ANALOG LINE INPUTS

Type	Unbalanced, low-noise input circuitry
Level	Consumer: -10 dBV nominal, 6.4 dBV maximum
Frequency Response	20 Hz - 20 kHz: +0.20/-0.10 dB
THD + N	-100 dB (.001%) 1kHz at -1 dBFS
SNR	111 dB (A-weighted 22kHz BW)
Dynamic Range	111 dB (1kHz, A-weighted, 22kHz BW)
Channel Crosstalk	< -120 dB, (1 kHz signal at -1 dBFS)
Input Impedance	3.3K ohm

ANALOG LINE OUTPUTS

Type	Unbalanced, low-noise circuitry
Level	Consumer: -10dBV nominal, 6.4dBV maximum
Frequency Response	+0.05/-0.10 dB, (20 Hz - 20 kHz)
THD + N	-100 dB (.001%) 1kHz signal at -1 dBFS
SNR	116 dB (A-weighted, 22 kHz BW)
Dynamic Range	116 dB (1 kHz, A-weighted, 22 kHz BW)
Stereo Crosstalk	< -109 dB, (1 kHz signal at -1 dBFS)
Output Impedance	560 ohms

DIGITAL I/O

- S/PDIF**
- 2 in/2 out coaxial (transformer coupled output)
 - 2 in/2 out optical
 - AES/EBU or S/PDIF (switchable under software control)
- MIDI** 1 in, 1 out (16 MIDI channels)

SYNCHRONIZATION

- Internal Crystal Sync:** 44.1 kHz, 48 kHz, 96 kHz
 S/PDIF (optical or coaxial)
 Word Clock (sync card only) - (75 ohm termination, switchable)

- RMS JITTER @ 44.1K** SRSync SourceRMS jitter in picoseconds
 (Measured via Audio Precision 2) 44.1 Internal Crystal 596ps
 44.1 Optical Input 795ps

SYNC CARD

- SMPTE** Converts to/from longitudinal time code (LTC) to MIDI time code (MTC) and vice-versa
- 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

Dimensions & Weight

0404 PCI Card

- Weight:** 0.25lb / 0.10kg
- Dimensions:** **L:** 156mm **H:** 107mm

Sync 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=d866b60193933eb726660e7bd90dfb27>
- 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>

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.

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