

Audio Express™

User Guide for Windows

MOTU

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This equipment has been type 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 instruction manual, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause interference to radio or television equipment reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by any combination of the following measures:

- Relocate or reorient the receiving antenna
- Increase the separation between the equipment and the receiver
- Plug the equipment into an outlet on a circuit different from that to which the receiver is connected

If necessary, you can consult a dealer or experienced radio/television technician for additional assistance.

PLEASE NOTE: only equipment certified to comply with Class B (computer input/output devices, terminals, printers, etc.) should be attached to this equipment, and it must have shielded interface cables in order to comply with the Class B FCC limits on RF emissions.

WARNING: changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.



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Quick Reference: Audio Express Front Panel



- These two Precision Digital Trim™ rotary encoders provide triple-function control over the trim level, mix volume, and pan of the XLR/TRS combo jacks on the rear panel. Push the MIX knob (4) repeatedly to cycle among the four separate mixes and trim mode. The LEDs (12) indicate the current mix, or trim mode. When a mix is active, push and hold the MIX knob (4) to toggle between volume and pan control, as indicated by the LEDs (12).
When in trim mode, use the knob and input level meters (9) to calibrate the input signal level. These knobs provide +60 dB and +22 dB of boost, respectively, for the mic (XLR) and instrument (TRS) input jacks. Adjustment can be made in approximately 1 dB increments. Both jacks have preamps, so you can plug in just about anything: a microphone, a guitar or even a synth. For +4 dB signals, use the TRS line inputs instead. For the Mic (XLR) input, push the encoder to toggle a -20 dB pad; push and hold to toggle 48V phantom power.
When in mix mode, use the knob to control the jack's input volume or pan (as indicated by the LEDs; 12) for the currently selected mix.
- These two Precision Digital Trim™ rotary encoders provide triple-function control over the trim level, mix volume, and pan of the TRS line in 3-4 jacks on the rear panel. Push the MIX knob (4) repeatedly to cycle among the four separate mixes and trim mode. The LEDs (12) indicate the current mix, or trim mode. When a mix is active, push and hold the MIX knob (4) to toggle between volume and pan control, as indicated by the LEDs (12).
When power is off, push the PHONES knob to power on the Audio Express; push and hold to turn it off. When connected to the computer via FireWire, the Audio Express is powered by its FireWire connection. When connected via USB, it must be powered with the included DC power adapter.
- This rotary encoder controls the input volume and pan of the S/PDIF jack for the currently selected mix. In trim mode, this knob provides 22 dB of gain in approximately 1 dB increments.
- This multi-function knob controls volume of the Audio Express's four mix buses (12). Push the knob repeatedly to cycle among the four mix buses, as indicated by the LEDs (12). When a mix is active, push and hold the VOL knob to toggle the individual channel knobs (1, 2 and 3) between volume and pan control, as indicated by the LEDs (12).
- Turn the PHONES knob to control the headphone jack volume. Push the knob to also control the Audio Express's main outs, as indicated by the "+main" LED (6). Push again to control *only* the main outs (instead of phones); in this case, the "+main" LED flashes.
When power is off, push the PHONES knob to power on the Audio Express; push and hold to turn it off. When connected to the computer via FireWire, the Audio Express is powered by its FireWire connection. When connected via USB, it must be powered with the included DC power adapter.
- The "+main" LED has three states: off, on, and flashing. Push the PHONES knob to cycle among three volume control modes: phone jack only (off), phones+main outs (on), or main outs only (flashing).
- These LEDs indicate whether the -20 dB pad or 48V phantom power is enabled or disabled for the corresponding mic input (1).
- The MAIN OUT meters provide five-segment stereo metering for the main outs ranging from -42 dB to -1 dB. The LINE meters provide four-segment metering for the line inputs 3-4 and signal activity LEDs at the bottom for line outs 3-4.
- The MIC / INST meters provide ten-segment ladder LED metering for the two mic/instrument combo jacks, ranging from -42 dB to clip.
- Indicates the current operational sample rate.
- The MIDI and S/PDIF activity LEDs indicate signal presence for each input and output.
- The Mix LEDs indicate what is being controlled by the mix/trim knobs (1, 2, 3 and 4). Push the MIX knob (4) repeatedly to cycle among the four separate mixes and trim mode. When a mix is active, the knobs control input levels for the current mix. Push and hold the MIX knob (4) to toggle between volume and pan control. When trim mode is selected, the knobs control input trim levels.
- This is a standard quarter-inch stereo headphone jack. From the factory, its output matches the main outs on the rear panel. But it can be programmed to mirror any other output pair (digital or analog). It can even be programmed to serve as its own independent output. See "Phones Assign" on page 30. Use the volume knob above to control its level.

Quick Reference: Audio Express Rear Panel



1. Connect a MIDI device here using standard MIDI cables. Connect the Audio Express MIDI OUT port to the MIDI IN port on the other device. Conversely, connect the Audio Express MIDI IN port to the MIDI OUT port on the other device. You can connect different devices to each port, such as a controller device to the IN port and a sound module to the OUT port. You can also daisy-chain MIDI devices, but be sure to manage their MIDI channels (so that they don't receive or transmit on the same channel).
2. These jacks provide stereo, 24-bit S/PDIF digital input and output at any sample rate up to 96kHz.
3. This jack accepts any standard 9-18V DC power supply with either tip-positive or tip-negative polarity.
4. These two balanced, quarter-inch jacks serve as the Audio Express's main outputs. You can connect them to a set of powered studio monitors and then control the volume from the front panel volume knob. To hear disk tracks in your audio software on these main outputs, assign

5. The Audio Express's analog line outputs are gold-plated, balanced +4dB TRS (tip/ring/steeve) quarter-inch connectors that can also accept an unbalanced plug. They are equipped with 24-bit, 128x oversampling converters.
6. These XLR/TRS combo jacks accept either a mic cable or a cable with a quarter-inch plug. Use the front panel mic 1-2 encoders to apply up to +60 dB or +22 dB of boost, respectively, for the mic (XLR) or instrument (TRS) input. To toggle the -20 dB pad for the XLR (mic) input, push the corresponding front-panel mic encoder. To toggle 48V phantom power for the XLR (mic) input, push and hold the corresponding front-panel encoder.

7. Equipped with 24-bit 96 kHz converters, these two analog inputs are gold-plated, balanced TRS (tip/ring/steeve) quarter-inch connectors that can also accept an unbalanced plug. They do not have microphone preamps, so they are best used for synthesizers, drum machines, effects processors, and other instruments with line level signals (either -10 dB or +4 dB). These inputs are also equipped with the Audio Express's Precision Digital Trim™ feature: digitally controlled analog trims that let you adjust input level in approximately 1 dB increments from either front panel or the included CueMix FX software. The trim can be adjusted over a range of 22 dB.

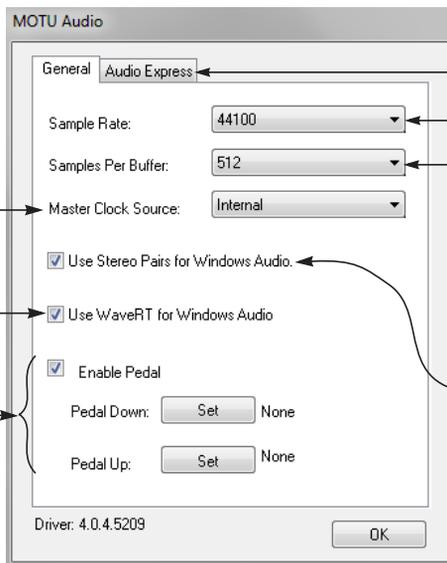
8. Connect a standard foot switch here for hands-free punch-in and punch-out during recording. For details about how to set this up, see "Enable Pedal" on page 30.
9. Connect the Audio Express to the computer here via either FireWire or USB2, using either the standard 1394 FireWire A or USB cable provided with your Audio Express. There's not much difference, except that FireWire offers bus-powered operation (without the DC power supply). **Important note: it is best to turn off the Audio Express when plugging in the FireWire cable, as this avoids the possibility of static discharge, which can harm the electrical components in the Audio Express or your computer.**

Quick Reference: MOTU Audio Console

Determines the clock source for your Audio Express. If you're just using the analog ins and outs, set this to *Internal*. The other settings are for digital transfers via S/PDIF or synchronization to time code or other audio devices.

Check this option if the audio software you are using with the Audio Express does not support Windows WDM drivers and instead only supports legacy MME (Wave) drivers. When checked, this option makes all of the Audio Express inputs and outputs available to legacy Wave-driver compatible audio software.

If you have a foot switch connected to the Audio Express, these settings let you map the foot switch to any computer keyboard key for both the up and down position. For details about how to set this up, see "Enable Pedal" on page 30.



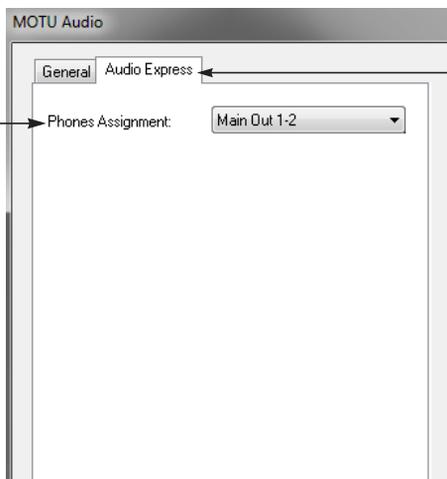
Click the tabs to access general MOTU interface settings or settings specific to the Audio Express (or other connected interface).

Choose the global sample rate for the system here.

Choosing a smaller setting here reduces the delay you may hear when listening to live input that you are running through effects plug-ins in your software. But lower settings also increase the strain on your computer. For details, see "Samples Per Buffer" on page 29.

This option should be enabled (checked) in most cases. For details, refer to "Use stereo pairs for Windows audio" on page 30.

This menu lets you choose what you will hear from the headphone jack. To mirror the main outs, choose *Main Out 1-2*. Or you can mirror any other output pair. To hear the phones as their own independent output, choose *Phones 1-2*.



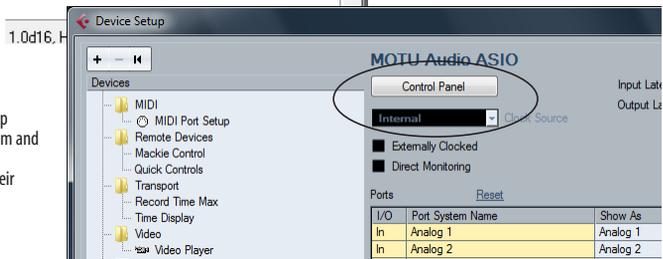
Click the Audio Express tab to access these settings.

Device Setup in Cubase

How to access these settings

There are several ways to access these settings:

- From the Windows Start menu, choose *Programs>MOTU>MOTU Audio Console*.
- From within Cubase, go to the Device Setup window, click the MOTU Audio ASIO list item and click the *Control Panel* button.
- From within other applications, refer to their documentation.





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OVERVIEW

The Audio Express is a hybrid FireWire and USB2 audio interface for Mac and Windows that offers six independent inputs and six independent outputs. Both analog and digital I/O are offered at sample rates up to 96 kHz. All inputs and outputs can be accessed simultaneously. The Audio Express is housed in a sturdy, compact half-rack enclosure that connects directly to a computer via a standard FireWire or USB cable.

The Audio Express offers the following main features:

- Universal computer connectivity via FireWire or high-speed USB 2.0
- Bus-powered operation (FireWire only)
- Two 24-bit analog quarter-inch (TRS) inputs
- Four 24-bit analog quarter-inch (TRS) outputs
- Two combo XLR/TRS mic/guitar inputs with preamps, 48V phantom power, 20 dB pad, and Precision Digital Trim™ preamp gain adjustment
- Operation on all I/O at standard sample rates up to 96 kHz
- Digitally controlled analog trim for all analog inputs
- Coaxial S/PDIF digital I/O at sample rates up to 96 kHz
- MIDI I/O

- Foot switch for hands-free punch-in/out
- On-board SMPTE synchronization
- Headphone jack with independent output and volume control
- Front panel volume control of the Main Outs
- Front-panel mixing of live inputs
- Front panel metering and status LEDs
- Stand-alone operation
- Mac and Windows drivers for multi-channel operation and across-the-board compatibility with any audio software on current Mac and Windows systems
- CueMix FX cross-platform mixing software with attractive graphic mixing and a convenient tabbed interface for quick access to all features in one window. CueMix also provides a full-screen real-time FFT display, spectrogram “waterfall” display, oscilloscope, X-Y plot, linear or polar phase analysis, and a tuner.

With a variety of I/O formats, mic preamps and no-latency mixing and processing of live input, the Audio Express is a complete, portable “studio in a box” when used with a Mac or Windows computer.

THE AUDIO EXPRESS REAR PANEL

The rear panel has the following connectors:

- Four gold-plated, balanced +4dB quarter-inch (TRS) analog outputs (with 24-bit 96 kHz converters)
- Two gold-plated, balanced quarter-inch (TRS) analog inputs (with 24-bit 96 kHz converters)
- Two combo XLR/TRS mic/instrument inputs

- Coaxial S/PDIF in/out
- MIDI IN and MIDI OUT connectors
- 1394 FireWire A connector
- High-speed USB 2.0 connector
- Foot pedal input
- DC power jack

6 inputs and 6 outputs

All Audio Express inputs and outputs can be used simultaneously, for a total of 6 inputs and 6 outputs:

Connection	Input	Output
Analog 24-bit 96 kHz on bal/unbal TRS	2	4
Mic preamps 24-bit 96 kHz on XLR/TRS combo	2	-
SPDIF 24-bit 96kHz digital	2	2
Total	6	6

All inputs and outputs are discrete and can be active simultaneously.

The headphone outputs can operate as an independent output pair, or they can mirror any other Audio Express output pair, such as the main outs.

Mic/guitar inputs with preamps

The two mic/instrument inputs are equipped with preamps and “combo” XLR/TRS jacks, which accept XLR microphone inputs or quarter-inch guitar/instruments inputs. Individual 48 volt phantom power and a 20 dB pad can be supplied independently to each mic input. The Precision Digital Trim™ knobs on the front panel for each mic/instrument input provide 60 dB of boost for the XLR mic input and 22 dB of boost for the TRS instrument input in approximately 1 dB increments.

Analog

The quarter-inch analog inputs are equipped with 24-bit 96 kHz A/D converters. The analog outputs have 24-bit 96 kHz D/A converters. All audio is carried to the computer in a 24-bit data stream.

All quarter-inch analog outputs and inputs are on balanced TRS +4dB jacks. All of these jacks can also accept unbalanced plugs.

Precision Digital Trim™

All of the Audio Express’s analog inputs are equipped with digitally controlled analog trims that allow adjustments in approximately 1 dB increments. The mic/guitar input trims can be adjusted using front-panel digital rotary encoders with 60 dB of adjustment for the mic input and 22 dB for the TRS input. All analog inputs, including the rear-panel TRS analog inputs, can be trimmed using the Audio Express’s included CueMix FX control software for Mac and Windows. This allows you to fine-tune trim settings for synths, effects modules and a wide variety of analog inputs for optimum levels. Different trim configurations can then be saved as files on disk for instant recall.

S/PDIF

The Audio Express rear panel provides coaxial S/PDIF input and output.

MIDI I/O

The Audio Express’s standard MIDI IN and MIDI OUT jacks supply 16 channels of MIDI I/O to and from the computer via the Audio Express’s FireWire or USB connection.

Pedal input

The quarter-inch foot pedal jack accepts a standard foot switch. When you push the foot switch, the Audio Express triggers a programmable keystroke on the computer keyboard. For example, with MOTU’s Digital Performer audio sequencer software, the foot switch triggers the 3 key on the numeric keypad, which toggles recording in

Digital Performer. Therefore, pressing the foot switch is the same as pressing the 3 key. The Audio Express Control Panel software lets you program any keystroke you wish.

On-board SMPTE synchronization

The Audio Express can resolve directly to SMPTE time code via any analog input, without a separate synchronizer. It can also generate time code via any analog output. The Audio Express provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy.

The included MOTU SMPTE Console™ software provides a complete set of tools for generating and regenerating SMPTE time code, which allows you to slave other devices to the computer. Like CueMix FX, the synchronization features are cross-platform and compatible with any audio software that supports the ASIO2 sample-accurate sync protocol.

Hybrid FireWire/USB2 connectivity

FireWire has long been recognized as a reliable, high-performance connectivity standard for professional MOTU audio interfaces. Meanwhile, high-speed USB2 has also developed into a widely adopted standard for connecting peripheral devices to personal computers.

To fully support both formats, your Audio Express Hybrid audio interface is equipped with both a FireWire A (400 Mbit/sec) connector and a high-speed USB2 (480 Mbit/sec) connector, and you can use either port to connect the Audio Express to your computer. This gives you maximum flexibility and compatibility with today's ever-expanding universe of Mac and Windows computers.

The Audio Express has the ability to power itself from its FireWire connection to the computer, for convenient, mobile bus-powered operation.

Power supply

If you do not want the Audio Express to draw power from the computer, and AC power is available, you can power the Audio Express from any standard 9-18V, 5 watt DC power supply with any polarity (tip positive or negative).

THE AUDIO EXPRESS FRONT PANEL

Front-panel mixing

The six digital rotary encoders on the Audio Express front panel provide hands-on mixing of all live inputs. In fact, these controls provide control of up to four separate mixes. Use the MIC, LINE and S/PDIF knobs to control input volume for each mix; push the MIX knob to cycle among the four different mixes, plus trim mode, which lets you adjust input trim for each across all mixes.

48V phantom power and pad

The two MIC knobs let you toggle independent 48V phantom power and -20 dB pad settings for the two mic inputs. See “Mic/guitar inputs with preamps” on page 10.

Headphone output and main volume control

The Audio Express front panel provides a quarter-inch stereo headphone output jack and volume knob, which can be configured to control phones volume, phones and main outs together, or the main outs by themselves.

Status and metering LEDs

The LED section provides visual indication of audio levels, audio activity status, and the current settings for all front-panel controls.

16-BIT AND 24-BIT RECORDING

The Audio Express system handles all data with a 24-bit signal path, regardless of the I/O format. You can record and play back 16-bit or 24-bit audio files at any supported sample rate via any of the Audio Express's analog or digital inputs and outputs. 24-bit audio files can be recorded with any compatible host application that supports 24-bit recording.

CUEMIX MIXING AND EFFECTS

All Audio Express inputs and outputs can be routed to the on-board CueMix 8-bus (4 stereo) digital mixer driven by hardware-based DSP. The mixer allows you to mix all inputs to any output pair directly in the Audio Express hardware, independent of the computer. The on-board mixer can be used when the Audio Express is operating stand-alone (without a computer) as a complete portable mixer.

HOST AUDIO SOFTWARE

The Audio Express system ships with standard Windows drivers that allow you to record, edit, play back and mix your Audio Express projects using your favorite Windows audio software.

CHAPTER 2 **Packing List and Windows System Requirements**

PACKING LIST

The Audio Express ships with the items listed below. If any of these items are not present in your Audio Express box when you first open it, please immediately contact your dealer or MOTU.

- One Audio Express I/O rack unit
- One 6-pin to 6-pin IEEE 1394 “FireWire” cable
- One USB cable
- One set of removable rack ears
- One Audio Express Mac/Windows manual
- One cross-platform CD-ROM
- Product registration card

WINDOWS SYSTEM REQUIREMENTS

The Audio Express system requires the following Windows system:

- Pentium 4 CPU (or equivalent) 1 GHz or faster
- 1 GB of RAM; 2 GB or more is recommended
- Available FireWire or USB 2.0 port
- A large hard drive (preferably at least 250 GB)
- Windows 7 or Vista

PLEASE REGISTER TODAY!

Please register your Audio Express today. There are two ways to register.

- Visit www.motu.com/registration to register online

OR

- Fill out and mail the included product registration card

As a registered user, you will be eligible to receive technical support and announcements about product enhancements as soon as they become available. Only registered users receive these special update notices, so please register today!

Thank you for taking the time to register your new MOTU products!

IMPORTANT! Run the Audio Express Software Installer First

OVERVIEW

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- MOTU MIDI Driver 16
- CueMix FX 16
- MOTU SMPTE Console 16

INSTALL THE AUDIO EXPRESS SOFTWARE FIRST!

Before you connect the Audio Express audio interface to your computer and turn it on, insert the Audio Express software CD and run the Audio Express Software Installer. This ensures that all the Audio Express components are properly installed in your system.

If Windows asks you to locate the drivers

If you've already connected the Audio Express to your computer and switched it on, Windows probably issued an alert notifying you that the Audio Express requires drivers, followed by another window asking you to locate the drivers on disk. If this happens:

- 1 Cancel the driver search.
- 2 Switch off the Audio Express.
- 3 Run the Audio Express Software Installer as instructed in the next section.

INSTALLING THE AUDIO EXPRESS SOFTWARE

To install the Audio Express software, insert the MOTU Universal Audio Installer CD and follow the directions it gives you on your computer screen. The Audio Express ships with the following software and drivers for Windows:

Software component	Purpose
MOTU Audio Console	Provides access to all of the settings in the Audio Express hardware.
CueMix FX	Gives you complete control over the 896mk3's CueMix FX on-board mixer, which provides no-latency monitoring, mixing and processing of live inputs through your Audio Express.
MOTU SMPTE Setup	Provides access to the Audio Express system's SMPTE time code sync features.
MOTU ASIO Driver	Allows Cubase or other ASIO-compliant software to do multi-channel input and output with the Audio Express. Only required if you are using Cubase or another ASIO-dependent program.
MOTU WDM Driver	Allows any WDM-driver compatible audio software to do multichannel input and output with the Audio Express.
MOTU MIDI Driver	Provides MIDI input and output via the Audio Express MIDI ports.

MOTU AUDIO CONSOLE

MOTU Audio Console (available in the Start menu) gives you access to all of the settings in the Audio Express, such as the clock source and sample rate. For complete details, see chapter 10, “CueMix FX” (page 51).

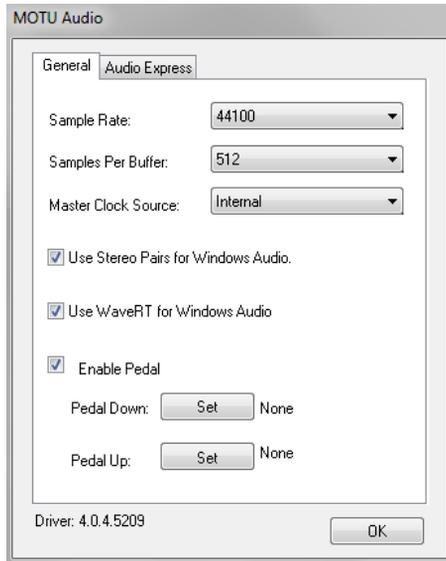


Figure 3-1: MOTU Audio Console gives you access to all of the settings in the Audio Express hardware.

ASIO MOTU AUDIO DRIVER

ASIO stands for *Audio Streaming Input and Output*. The ASIO MOTU FireWire driver allows the Audio Express to provide multi-channel input and output for Steinberg’s Cubase and Nuendo software, or any other audio application that supports ASIO drivers.

The MOTU ASIO audio driver is only required if you are using Cubase (or another audio program that relies on the ASIO driver to support multi-channel I/O with the Audio Express).

The MOTU ASIO audio driver is installed by the Audio Express Software Installer and properly registered with Windows, so you don’t need to be concerned about its installation or location.

For details about using Cubase and Nuendo with the Audio Express, see chapter 7, “Cubase, Nuendo, Live and Other ASIO Software” (page 35).

THE MOTU AUDIO WDM DRIVER

The MOTU Audio WDM driver provides standard multi-channel input and output for audio applications running under Windows XP or Vista. See chapter 8, “SONAR and other WDM Software” (page 41) for details.

The MOTU Audio installer CD installs the MOTU Audio WDM driver into Windows for you.

MOTU MIDI DRIVER

This driver allows you to access the Audio Express’s MIDI input and output ports. The ports are published in Windows and are available to all MIDI software.

CUEMIX FX

This program provides a mixing console that gives you control over the Audio Express’s no-latency CueMix FX on-board mixing and effects processing. For details, see chapter 10, “CueMix FX” (page 51).

MOTU SMPTE CONSOLE

The MOTU SMPTE Console software provides a complete set of tools for resolving the Audio Express to SMPTE time code, and to generate SMPTE for striping, regenerating or slaving other devices to the computer. For details, see chapter 11, “MOTU SMPTE Console” (page 75).

CHAPTER 4 **Installing the Audio Express Hardware**

OVERVIEW

Here's an overview for installing the Audio Express:

Important note before you begin! 17

Take these precautions to prevent damage to your computer, the Audio Express and other equipment.

Connect the Audio Express interface 18

Connect the Audio Express to the computer.

Connect audio inputs and outputs 20

Make analog and digital connections as desired.

Connect MIDI gear 21

Connect a controller, synth or control surface.

Connect and sync S/PDIF devices 22

Connect a DAT deck, effects processor or other device with digital I/O, but be sure to make the correct clock source settings.

Power options 22

Choose from among several convenient options.

A typical Audio Express setup 24

An example setup for computer-based mixing/FX.

Connect multiple MOTU interfaces 25

Connect additional Audio Express or other audio interfaces.

IMPORTANT NOTE BEFORE YOU BEGIN!

Before you begin installing the Audio Express (or any bus-powered device), take these important precautionary measures to avoid damaging the sensitive electrical components in your computer, the Audio Express or other devices being connected:

- Turn off the computer.
- Turn off the Audio Express (push and hold the phones volume knob).
- Turn off the power of any other devices.
- Touch the metal casing of the Audio Express to discharge any static electricity that you may be carrying just before the installation.

After you have made all of the necessary connections, as described in this chapter, turn on the devices in this sequence:

1. Turn on the computer.
2. Turn on the Audio Express.
3. Turn on other devices connected to the Audio Express.

CONNECT THE AUDIO EXPRESS INTERFACE

Your Audio Express audio interface is equipped with both a FireWire A (400 Mbit/sec) connector and a hi-speed USB2 (480 Mbit/sec) connector, and you can use either port to connect the Audio Express to your computer. This gives you maximum flexibility and compatibility with today's ever-expanding universe of Mac and Windows computers.

Which should I use: FireWire or USB2?

If your computer does not have a FireWire port, then obviously you will need to connect the Audio Express to one of its high-speed USB 2.0 ports.

If your computer has both FireWire and USB2, then it is your choice, and your decision may depend mostly on other peripherals you may also have, or bus power, as explained below.

Bus-powered operation requires FireWire

There is only one significant difference between FireWire and USB2 operation: bus power. FireWire provides enough power on the FireWire bus that the Audio Express can be powered solely by its FireWire connection to the computer. For complete information, see "Bus power requirements" on page 22.

If you use USB2, you must also use the DC power supply included with your Audio Express, as USB2 does not supply enough power by itself.

If you are connecting via FireWire

- 1 Before you begin, make sure your computer and the Audio Express are switched off.
- 2 Plug one end of the Audio Express FireWire cable (included) into the FireWire socket on the computer as shown below in Figure 4-1.
- 3 Plug the other end of the FireWire cable into the Audio Express I/O as shown below in Figure 4-1.



Figure 4-1: Connecting the Audio Express to the computer via FireWire.

Make absolutely sure to align the flat side of the FireWire plug properly with the flat side of the FireWire socket on the Audio Express. If you attempt to force the plug into the socket the wrong way, you can damage the Audio Express.

High Speed USB 2.0 versus USB 1.1

There are primarily two types of USB host controllers widely available on current personal computers. USB 1.1 controllers support simple peripherals that don't require a high speed connection, such as a computer keyboard, a mouse, or a printer. USB 2.0 controllers support high speed devices such as the Audio Express. Since the Audio Express requires a high speed connection, it must be connected to a USB 2.0 host controller or hub.

For the most reliable connection, it is recommended that you connect the Audio Express directly to one of your computer's USB 2.0-compatible ports. However, since USB 2.0 hubs are compatible with both types of devices, the Audio Express can be connected to a USB 2.0 hub along with USB 1.1 devices if necessary. The Audio Express will not operate properly if it is connected to a USB 1.1 hub.

Follow these instructions to determine whether your computer supports USB 1.1 or USB 2.0:

- 1 Open the Windows Control Panel, and choose Hardware and Sound.
- 2 In the Devices and Printers section, click Device Manager.
- 3 In Device Manager, open the Universal Serial Bus Controllers section.
- 4 Look in the list of USB devices. An Enhanced USB Host Controller Interface (EHCI) represents a USB 2.0 controller. An Open USB Host Controller Interface (OHCI) or Universal USB Host Controller Interface (UHCI) represents a USB 1.1 controller.

If you are connecting via high-speed USB 2.0

- 1 Before you begin, make sure your computer and the Audio Express are switched off.
- 2 Plug the flat "type A" plug of the Audio Express USB cable (included) into a USB2-equipped socket on the computer as shown below in Figure 4-1.
- 3 Plug the squared "type B" plug of the USB cable into the Audio Express I/O as shown below in Figure 4-1.



Figure 4-2: Connecting the Audio Express to the computer via USB.

Switching Between FireWire and USB

Most of the time, the Audio Express can detect whether a FireWire or a USB cable is connected to it, and it will automatically switch to FireWire operation or USB operation accordingly. Some FireWire or USB ports that do not provide bus power can not be automatically detected by the Audio Express, and you may need to switch the device to FireWire mode or USB mode manually:

To switch to	Press and hold this knob while turning on the Audio Express
FireWire	LINE 3 knob
USB	LINE 4 knob

When the Audio Express switches to FireWire operation, it displays *F* on the front panel LEDs. When it switches to USB operation, it displays *U* on the front panel LEDs.

CONNECT AUDIO INPUTS AND OUTPUTS

Here are a few things to keep in mind as you are making audio connections to other devices.

Mic/instrument inputs

Connect a microphone, guitar, instrument or other analog input to the XLR/TRS combo jack with either a standard mic cable or a balanced cable with a quarter-inch plug.

🔊 Do not connect a +4 (line level) XLR cable to the mic inputs (because of the preamps). Use the separate quarter-inch inputs instead.

Phantom power

If you are connecting a condenser microphone or other device that requires phantom power, enable phantom power as follows:

- 1 Push the MIX knob repeatedly until the green *trim* LED is illuminated on the front panel.
- 2 Push and hold the corresponding MIC knob for a few seconds to toggle phantom power.

The red 48V LED will turn on or off accordingly.

Trim

The XLR mic input and the TRS instrument input are equipped with 60 dB and 22 dB of trim control, respectively. Use the digital trim encoders on the front panel to adjust the input level for each input:

- 1 Push the MIX knob repeatedly until the green *trim* LED is illuminated on the front panel.
- 2 Turn the MIC knob to adjust the trim.

The ten-segment MIC/INST meters provide visual feedback as you turn the knob. The Audio Express's input trims are digitally controlled, so they allow you to make fine-tuned adjustments in approximately 1dB increments. You can also adjust trim in the MOTU CueMix FX software. See “Input trim” on page 53.

20 dB pad

Each XLR mic input is equipped with a -20 dB pad. To toggle the pad for a mic input:

- 1 Push the MIX knob repeatedly until the green *trim* LED is illuminated on the front panel.
- 2 Push the corresponding MIC knob to toggle the pad. The amber pad LED turns on or off.

Combo jack summary

Use these general guidelines for the 48V phantom power, pad and trim settings on the two combo input jacks:

Input	48V	Pad	Trim
Condenser mic	On	As needed	As needed
Dynamic mic	Off	As needed	As needed
Guitar	Off	As needed	As needed

Quarter-inch analog

The quarter-inch analog inputs (3-4) and outputs (1-4) are balanced (TRS) connectors that can also accept an unbalanced plug.

The quarter-inch outputs are calibrated to produce a +4 dBu line level output signal.

Quarter-inch analog input trims

The quarter-inch inputs are calibrated to accommodate either +4 dBu or -10 dBV signals and are equipped with digitally controlled analog trims that provide +22 dB of gain. You can use either the front panel knobs or the included CueMix FX software to adjust the input trim. To adjust these trims using CueMix FX, see “Input trim” on page 53. To adjust the trims using the front panel knobs:

- 1 Push the MIX knob repeatedly until the green *trim* LED is illuminated on the front panel.
- 2 Turn the corresponding LINE knob to adjust the trim.

Main outs

In a standard studio configuration, the main outs are intended for a pair of studio monitors, but they can also be used as regular outputs for any purpose. Their reference level is +4dB. Use the front panel VOL knob to adjust the Main out volume. Push the VOL knob until you see *+main* LED turn red (or flash red) and then turn it to adjust the main output volume.

S/PDIF

If you make a S/PDIF digital audio connection to another device, be sure to review the digital audio clocking issues, as explained in “Connect and sync S/PDIF devices” on page 22.

The Audio Express on-board mixer also supplies 12dB of digital trim adjustment for the S/PDIF input pair, which can be adjusted from CueMix FX (“Input trim” on page 53) or the front panel (using the same procedure as explained earlier for “Quarter-inch analog input trims” above).

CONNECT MIDI GEAR

Connect your MIDI device’s MIDI IN jack to the Audio Express’s MIDI OUT jack (Connection A below). Conversely, connect the MIDI device’s MIDI OUT jack to the Audio Express’s MIDI IN jack (Connection B).

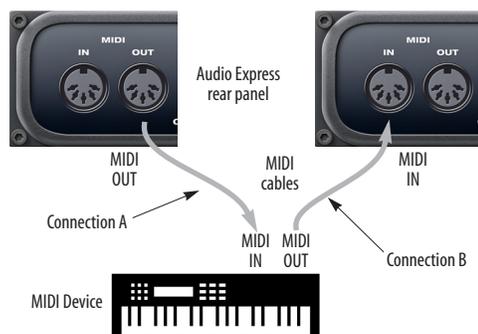


Figure 4-3: Connecting a MIDI device to the Audio Express.

One-way MIDI connections

MIDI devices that do not receive MIDI data, such as a dedicated keyboard controller, guitar controller, or drum pad, only need Connection B shown in Figure 4-3. Similarly, devices that never send data, such as a sound module, only need Connection A. Make both connections for any device that needs to both send and receive MIDI data.

Connecting additional gear with MIDI THRU

If you need to connect several pieces of MIDI gear, run a MIDI cable from the MIDI THRU of a device already connected to the Audio Express to the MIDI IN on the additional device as shown below in Figure 4-4. The two devices then share the Audio Express’s MIDI OUT port. This means that they share the same set of 16 MIDI channels, too, so try to do this with devices that receive on only one MIDI channel (such as effects modules) so their receive channels don’t conflict with one another.

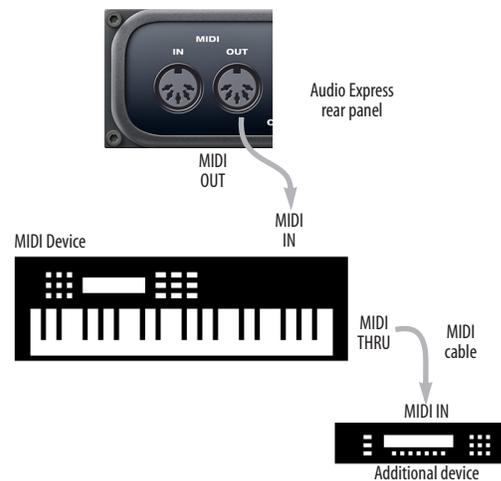


Figure 4-4: Connecting additional devices with MIDI THRU ports.

CONNECT AND SYNC S/PDIF DEVICES

DAT decks and other devices with S/PDIF digital I/O will sync to the Audio Express via the S/PDIF connection itself. Just connect it to the Audio Express via the S/PDIF connectors. When the device records a digital audio signal (from the Audio Express), it will simply synchronize to the clock provided by the digital audio input from the Audio Express.

When recording digital audio into your host audio software via the Audio Express's S/PDIF input, you can either resolve the external device to the Audio Express, as described above, or you can resolve the Audio Express to its S/PDIF input.



Figure 4-5: The setup for synchronizing a S/PDIF device with the Audio Express. Sync is achieved via the digital I/O connection itself. In this case, you have to choose S/PDIF as the Audio Express's clock source when resolving it to the other device.

POWER OPTIONS

The Audio Express can draw power from two possible sources:

1. the computer via FireWire, or
2. a DC power supply.

If you use USB2 to connect to your computer, you must also use the DC power supply included with your Audio Express, as USB2 does not supply enough power by itself.

Bus power requirements

The Audio Express draws all the power it needs from the FireWire bus connection to the computer. However, the FireWire connection to the computer must meet all of the requirements discussed below.

6-pin FireWire connectors

The Audio Express can only draw power over the FireWire bus from a 6-pin to 6-pin cable, or a 6-pin to 9-pin (FireWire B) cable. It cannot draw power from a FireWire cable with a 4-pin connector, as shown below:

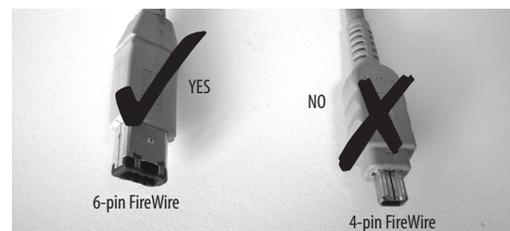


Figure 4-6: 4-pin FireWire connectors cannot be used for bus power.

When operating under bus power, daisy-chaining is not recommended

The Audio Express can be daisy-chained with other FireWire devices from a single FireWire connection to the computer. However, if the Audio Express is operating under bus power, this is not recommended. If you need to daisy chain the Audio Express with other devices on the same FireWire bus, power the Audio Express with the included power adapter. The other devices on the chain should also have their own power supply. In general, bus-powered FireWire devices should not be daisy-chained.

FireWire adapter products must be powered

If you are using a FireWire adapter (a third-party product that supplies one or more FireWire ports to your computer), it must have direct access to a power supply:

- **PCI FireWire cards** — If you plan to connect the Audio Express to a PCI card and run the Audio Express under bus power, the PCI card must have a

direct connection to the power supply harness inside your computer. This is the same power supply harness to which you connect internal hard drives, CD/DVD drives, etc.

■ **PCMCIA or ExpressCard slot adapters** — If you plan to connect the Audio Express to a PC card FireWire adapter (inserted in the PC card slot in your laptop), it must provide a 6-pin connection and it must also have its own power supply. Most commonly, these types of products have a DC power adapter. As you can see, however, this situation does not allow for remote battery operation, as the PC card adapter requires AC.

Examples of bus- powered operation

Here are a few typical examples of bus-powered Audio Express operation:

Bus power from a desktop computer

Your desktop computer is running off of its usual AC power connection, and the Audio Express draws power from the FireWire cable connected to the computer. There are no limits to running time.

Bus power from an AC-powered laptop

This scenario is identical to the desktop situation described above: the laptop is powered by AC, the Audio Express is powered via the FireWire bus and there are no limits to running time.

Bus power from a battery-powered laptop

The laptop is being powered by its own battery, and the Audio Express is being powered by its FireWire connection to the computer. So the laptop battery is supplying power to both the laptop and the

Audio Express. This is the most compact and portable operating scenario. Running time is determined by the capacity of the laptop battery. For extended recording sessions, bring extra, fully charged laptop batteries.

DC power supply

If you do not want the Audio Express to draw power from the computer, and AC power is available, you can power the Audio Express from any standard 9-18 volt, 5 watt DC power supply with any polarity (tip positive or negative), and amperage as shown below.

Voltage	Amperage
9 volts	1.33 amps
12 volts	1 amp
18 volts	0.66 amps

Turning off the Audio Express

To turn on the Audio Express, push the PHONES volume knob. To turn it off, push *and hold* the PHONES volume knob. When the Audio Express is turned off, it is really in a sort of “sleep” mode, where it still draws just enough power to detect the power switch (a digital encoder) when the Audio Express is turned back on. But the amount of power that the Audio Express draws when it is turned off is so small that it has very little practical impact. If you are running a laptop under battery power, and you are in a situation where you are not using the Audio Express and you need every last bit of laptop battery power, unplug the Audio Express entirely from the computer.

A TYPICAL AUDIO EXPRESS SETUP

Here is a typical Audio Express studio setup. In this example, no external mixer is needed. All mixing and processing can be done in the computer with

audio software, or you can use the Audio Express's CueMix™ no-latency mixer. You can control the Audio Express's four separate mixes from the front panel or from the included CueMix FX software.

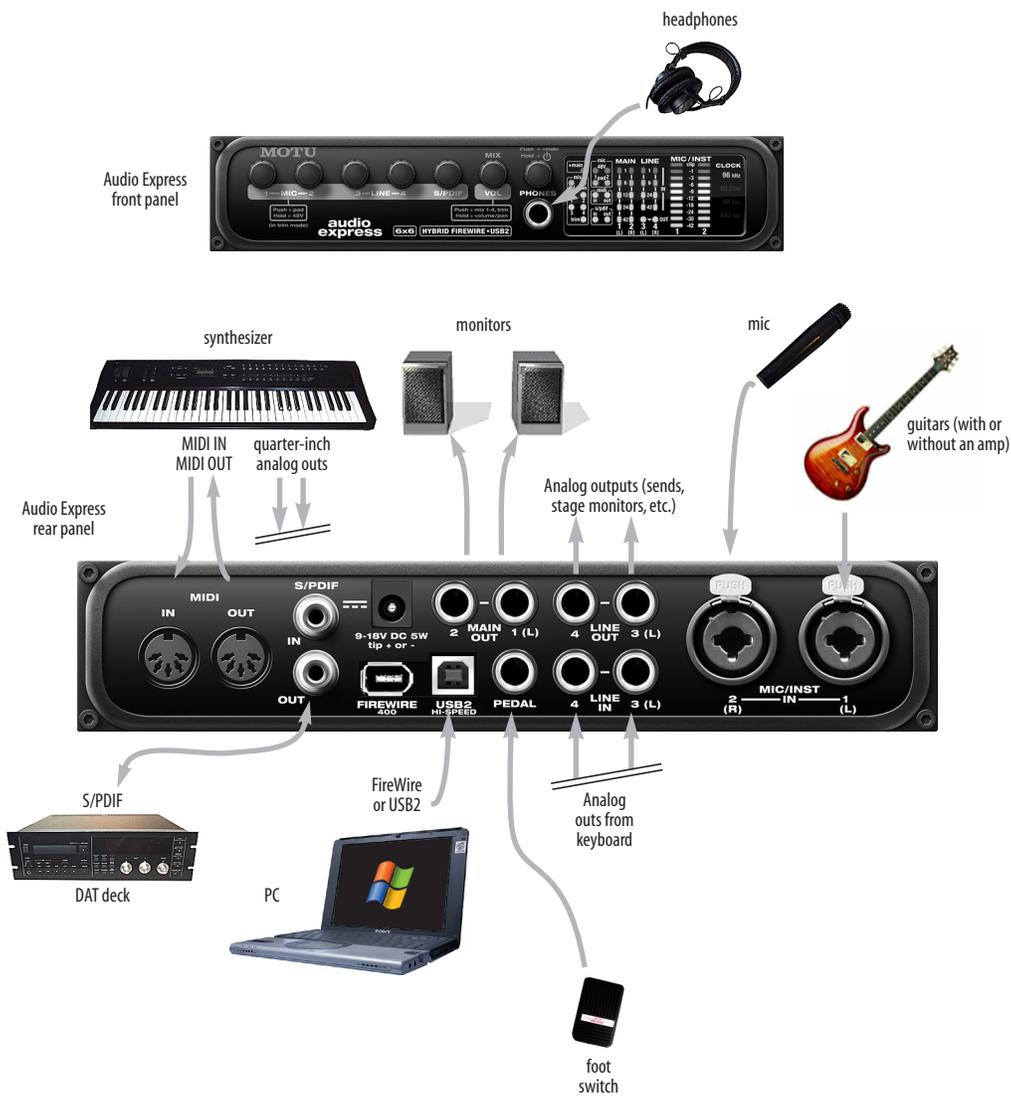


Figure 4-7: A typical Audio Express studio setup.

CONNECT MULTIPLE MOTU INTERFACES

You can daisy-chain up to four MOTU FireWire interfaces on a single FireWire bus, with the restrictions described in the following sections. Most computers have only one built-in FireWire bus (even if it supplies multiple FireWire sockets).

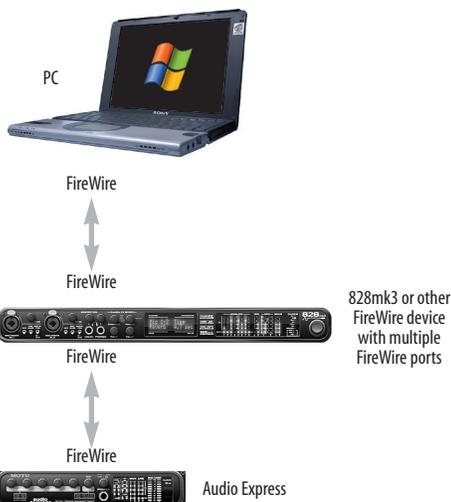


Figure 4-8: Connecting multiple MOTU FireWire audio interfaces.

Multiple interfaces cannot be bus-powered

Do not run the Audio Express under bus power when connecting it with other devices to the same FireWire bus. See “Power options” and “Bus power requirements” on page 22.

Multiple interfaces in MOTU Audio Console

MOTU Audio Console displays the settings for one interface at a time. To view the settings for an interface, click its tab as shown below in Figure 4-9.

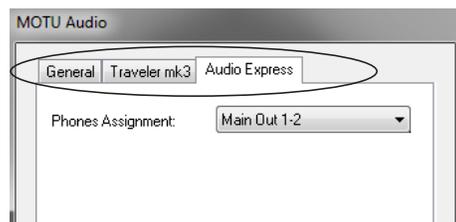


Figure 4-9: To view the settings for an interface, click its tab.

Synchronizing multiple interfaces

When multiple MOTU interfaces are connected, choose a *Master Clock Source* from the menu in the General tab (Figure 4-10). You can choose any available sync source from any connected interface (as shown in Figure 4-10), giving you a great deal of flexibility in choosing a clock source.

A separate *Clock* menu is provided for any interfaces not chosen as the master clock source, as demonstrated with the 828mk3 Hybrid in Figure 4-10. You can either resolve the interface to the *Master Clock Source* or to one of its own digital inputs or other sync sources. The latter scenario is useful for externally resolving the interface with the the interface chosen as the Master Clock Source.

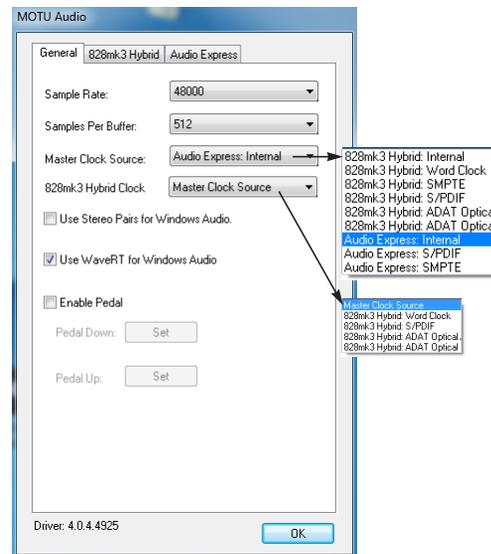


Figure 4-10: Choosing clock sources with multiple interfaces.

Connecting other MOTU FireWire interfaces

You can mix and match multiple Audio Express interfaces with other MOTU FireWire interfaces using a standard FireWire hub. You can daisy-chain 828mkIIs, 828mk3s, 896HDs, 896mk3s, Travelers, and Traveler-mk3s, which all have two FireWire ports convenient for daisy-chaining. Up to four interfaces can be combined on one FireWire bus.

Adding additional interfaces with a second FireWire bus

Third-party FireWire bus expansion products in the form of a cardbus (“PC card”) adaptor, ExpressCard adaptor or PCI card allow you to add

a second FireWire bus to your computer. It may be possible to add additional MOTU FireWire interfaces connected to such a third-party product, depending on the performance of the product and the performance of your host computer.

CHAPTER 5 MOTU Audio Console

OVERVIEW

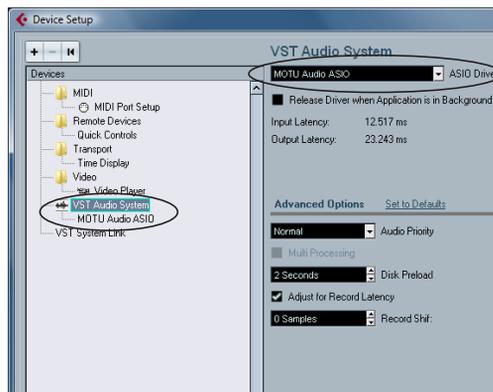
MOTU Audio Console gives you access to basic Audio Express hardware settings, such as sample rate, clock source and more.

Accessing the Audio Express settings	27
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Clock Source	28
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Enable Pedal	30
'Audio Express' tab settings	30
Phones Assign	30
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ACCESSING THE AUDIO EXPRESS SETTINGS

There are several ways to access MOTU Audio Console settings:

- From the Windows Start menu, choose *Programs>MOTU>MOTU Audio Console*
- In Cubase or Nuendo, open the Device Setup window, click *VST Audio System* and choose *MOTU Audio ASIO* from the *ASIO Driver* menu as shown below. Then click the *MOTU Audio ASIO* item in the list and click the *Control Panel* button.



- From within other ASIO-compatible programs, refer to their documentation.

General tab settings

The General tab provides settings that apply globally to all connected MOTU FireWire interfaces.

Audio Express tab settings

The Audio Express tab provides settings that apply to a specific Audio Express interface. If you have several Audio Express interfaces connected (or other MOTU interfaces), you'll see a separate tab for each one.

'GENERAL' TAB SETTINGS

Sample Rate

Choose the desired *Sample Rate* for recording and playback. The Audio Express can operate at 44.1 (the standard rate for compact disc audio), 48, 88.2, or 96 kHz. If you have a S/PDIF device connected to the Audio Express, make sure that it matches the Audio Express's sample rate.

☞ Mismatched sample rates cause distortion and crackling. If you hear this sort of thing, check the sample rate settings in your hardware and here in MOTU Audio Console.

Clock Source

The *Clock Source* determines the digital audio clock that the Audio Express will use as its time base. The following sections briefly discuss each clock source setting.

Internal

Use the *Internal* setting when you want the Audio Express to operate under its own digital audio clock. For example, you may be in a situation where all you are doing is playing tracks off hard disk in your digital audio software on the computer. In a situation like this, you most often don't need to reference an external clock of any kind.

Another example is transferring a mix to DAT. You can operate the Audio Express system on its internal clock, and then slave the DAT deck to the Audio Express via the S/PDIF connection (usually DAT decks slave to their S/PDIF input when you choose the S/PDIF input as their record source).

If you would like help determining if this is the proper clock setting for your situation, see "Connect and sync S/PDIF devices" on page 22.

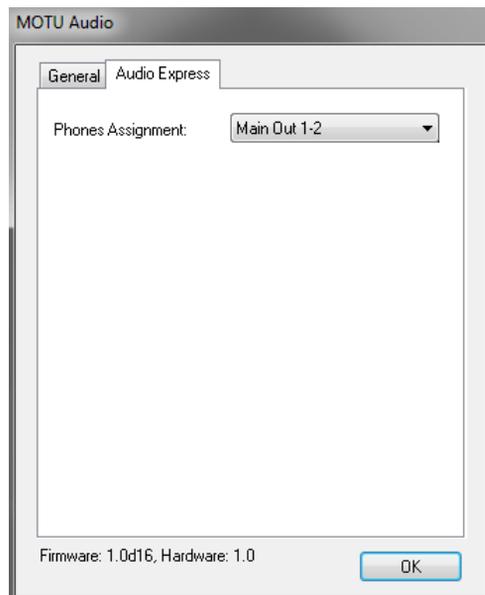
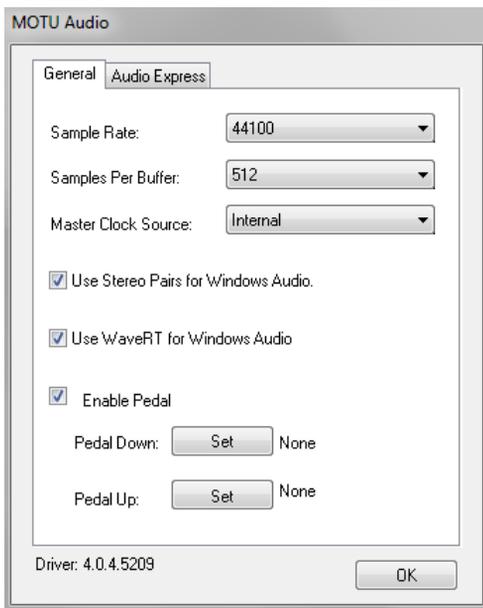


Figure 5-1: MOTU Audio Console gives you access to all of the settings in the Audio Express hardware.

S/PDIF

The *S/PDIF* clock source setting refers to the *S/PDIF* coaxial input jack on the Audio Express. This setting allows the Audio Express to slave to another *S/PDIF* device.

Use this setting whenever you are recording input from a DAT deck or other *S/PDIF* device into the Audio Express. It is not necessary in the opposite direction (when you are transferring from the Audio Express to the DAT machine).

For further details about this setting, see “Connect and sync *S/PDIF* devices” on page 22.

SMPTE

Choose this setting to resolve the Audio Express directly to SMPTE time code (LTC) being received via one of the Audio Express’s audio inputs. For details, see “Syncing to SMPTE time code” on page 78 and chapter 11, “MOTU SMPTE Console” (page 75).

Samples Per Buffer

The *Samples Per Buffer* setting lets you reduce the delay you hear when patching live audio through your audio software. For example, you might have a live microphone input that you would like to run through a reverb plug-in that you are running in your host audio software. When doing so, you may hear or feel some “sponginess” (delay) between the source and the processed signal. If so, don’t worry. This effect only affects what you hear: it is not present in what is actually recorded.

You can use *Samples Per Buffer* setting to reduce this monitoring delay—and even make it completely inaudible.

☛ If you don’t need to process an incoming live signal with software plug-ins, you can monitor the signal with no delay at all using CueMix FX, which

routes the signal directly to your speakers via hardware. For details, see chapter 10, “CueMix FX” (page 51).

Adjusting the *Samples Per Buffer* setting impacts the following things:

- The strain on your computer’s CPU
- The delay you hear when routing a live signal through your host audio software plug-ins
- How responsive the transport controls are in your software

This setting presents you with a trade-off between the processing power of your computer and the delay of live audio as it is being processed by plug-ins. If you reduce the *Samples Per Buffer*, you reduce patch thru latency, but significantly increase the overall processing load on your computer, leaving less CPU bandwidth for things like real-time effects processing. On the other hand, if you increase the *Samples Per Buffer*, you reduce the load on your computer, freeing up bandwidth for effects, mixing and other real-time operations. But don’t set the *Samples Per Buffer* too low, or it may cause distortion in your audio.

If you don’t process live inputs with software plug-ins, leave this setting at its default value of 1024 samples. If you do, try settings of 256 samples or less, if your computer seems to be able to handle them. If your host audio software has a processor meter, check it. If it starts getting maxed out, or if the computer seems sluggish, raise the *Samples Per Buffer* until performance returns to normal.

If you are at a point in your recording project where you are not currently working with live, patched-thru material (e.g. you’re not recording vocals), or if you have a way of externally monitoring input, choose a higher *Samples Per Buffer* setting. Depending on your computer’s CPU speed, you might find that settings in the middle work best.

The *Samples Per Buffer* setting also impacts how quickly your audio software will respond when you begin playback, although not by amounts that are very noticeable. Lowering the *Samples Per Buffer* will make your software respond faster; raising the *Samples Per Buffer* will make it a little bit slower, but barely enough to notice.

Monitoring live inputs without plug-in effects

As mentioned earlier, CueMix FX allows you to monitor dry, unprocessed live inputs with no delay at all. For complete details, see chapter 9, “Reducing Monitoring Latency” (page 45).

Use stereo pairs for Windows audio

Some legacy Windows audio applications require audio channels to be grouped in stereo pairs. If you are using old Windows audio software, and you are having issues accessing Audio Express channels from within the software, make sure to enable the *Use Stereo Pairs for Windows Audio* option. Newer software is also compatible with this option, so it is recommended that you always leave this option enabled.

Enable Pedal

Check the Enable Pedal option if a foot switch is connected to the Audio Express and you would like to trigger recording punch in/out (or other software functions) with it. Use the Set buttons to

determine what keystroke is triggered by the pedal-up and pedal-down positions. You can assign the pedal to any two keystrokes you wish. (You are not restricted to punch in/out.)

‘AUDIO EXPRESS’ TAB SETTINGS

Phones Assign

The *Phones Assign* setting lets you choose what you will hear from the headphone jack. Choose *Main Out 1-2* if you’d like the headphone output to match the main outs. If you choose *Phones 1-2*, this setting makes the headphone jack serve as its own independent output pair. You’ll see *Phones 1-2* as an additional audio destination in your host software’s audio output menus.

Disable Interface

You can disable an interface at any time with the *Disable interface* option. Doing so frees up the FireWire bandwidth required by the interface without turning it off. Switching off an interface accomplishes the same thing. To get MOTU Audio Console to forget about an interface entirely, you’ll see a *Forget* button in MOTU Audio Console. Just click the *Forget* button and MOTU Audio Console will no longer consider the interface to be present but off line (turned off).

CHAPTER 6 Audio Express Front Panel Operation

OVERVIEW

The Audio Express offers front-panel mixing via six rotary encoders and a bank of status LEDs.

- Push-button rotary encoders 31
- Power switch 31
- Phones 31
- Volume control for phones and main outs 31
- Mixing with the front panel controls 32
- Visual feedback when adjusting mix controls 34
- Metering and activity LEDs 34
- Restoring factory defaults 34

PUSH-BUTTON ROTARY ENCODERS

All of the knobs on the Audio Express front panel are push-button digital rotary encoders. In many cases, you can either push the knob, hold it in, or turn it to make a setting.

POWER SWITCH

Push the PHONES knob to power on the Audio Express. Push in and hold the knob to turn it off.

PHONES

From the factory, the PHONES jack (Figure 6-1) mirrors the main outs, but it can mirror any other output pair (digital or analog). See “Phones Assign” on page 30.

VOLUME CONTROL FOR PHONES AND MAIN OUTS

Push the PHONES knob repeatedly to cycle among three volume control modes, as indicated by the +main LED:

Volume control	+main LED
Phones only	Off
Phones + main out	On
Main out only	Flashing



Figure 6-2: When the red “+main” LED is on, the PHONES knob controls both the headphone out and the main outs.

As you turn the PHONES knob to adjust volume, the MIC/INST ladder LEDs (Figure 6-1) provide visual feedback.



Figure 6-1: The Audio Express front panel.

MIXING WITH THE FRONT PANEL CONTROLS

The Audio Express can operate as a mixer, and you can use the knobs on the front panel to control the mix. The mixer combines the signals of all inputs and sends the mixed signal to an output pair, such as the main outs. You can control the relative volume of the inputs, as well as the overall volume of the mix signal on output.

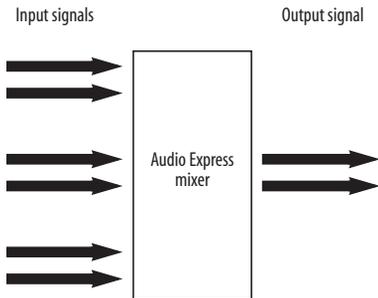


Figure 6-3: The Audio Express mixer takes all the input signals and combines them to an output. You control the volume of each input separately to produce a good mix.

Controlling the volume of inputs

Use the *Input controls* (Figure 6-1) to control the volume of the Audio Express's three input pairs. Notice that you can mix the digital S/PDIF input alongside the four analog inputs. Push to mute or unmute the input.

Controlling the volume of the mix

Use the *Mix volume* knob (Figure 6-1) to control the overall output volume of the mix.

Four separate mixes

The Audio Express provides a separate mix for each of its four output pairs, as follows:

Mix name	Physical output
Mix 1	Main Out 1-2
Mix 2	Analog 1-2
Mix 3	S/PDIF 1-2
Mix 4	Phones 1-2

This means that for each Audio Express output pair, the volume of each input can be completely different than it is for other output pairs. This allows you to create different mixes and send them to separate destinations. For example, you could send one mix to a PA system, and send a separate, different mix to stage monitors. Yet a third mix could be created for the headphones, and a fourth mix could be sent to the digital outputs to a recording device of some kind (or an effects processor).

Choosing a mix to control

The Audio Express front panel controls only one mix at a time. To choose which mix you are controlling, push the *MIX/VOL* knob (Figure 6-4) repeatedly. The *current mix* LEDs (Figure 6-4) indicate the currently selected mix.



Figure 6-4: The Mix LEDs indicate which mix you are controlling with the front panel.

Controlling pan

If a mix is selected and the volume (vol) LED is illuminated as shown in (Figure 6-4), this indicates that the front-panel knobs are currently controlling each input's mix volume. If you wish to control the left-right panning of the input, push *and hold* the *MIX/VOL* knob. Doing so causes the *pan* LED (Figure 6-4) to illuminate. You can now use the knobs to pan each mono input left or right. Like volume, the pan setting for the input can be different for each of the four mixes. Push the knob to toggle between pan-center or pan-left/right.

Use the following gestures for controlling pan for the (stereo) S/PDIF input, or any inputs that are configured as a stereo pair by the CueMix FX software (see the next section):

Do this	To do this
Turn	To adjust balance or width
Push and hold	To toggle between balance and width
In balance mode, push	To set balance to center
In width mode, push	To toggle between 100% (full stereo) and 0% (full mono)

To go back to controlling volume, push and hold the MIX/VOL knob again. The volume (*vol*) LED once again becomes illuminated.

Stereo pairs

From the factory, the two mic/instrument inputs and two TRS analog jacks are configured as individual mono inputs, each with its own knob. However, the CueMix FX software gives you the option of controlling them as stereo pairs. (See “Mono/stereo pairing” on page 53.) If you do so, the two front panel knobs that correspond to the pair will each affect both channels in identical fashion.

For example, if the two mic inputs are configured as a stereo pair, you could use either the MIC 1 knob or the MIC 2 knob to control the volume of the pair. Panning can also be controlled using either knob, as explained in the previous section for stereo pairs.

The S/PDIF input is different than the analog inputs because its two channels are, by default, grouped as a stereo pair. In addition, there is only one knob on the front panel for the pair. Most of the time, you will operate the S/PDIF input in stereo. However, the CueMix FX software does let you configure the S/PDIF channels as two separate mono channels. If you choose to do so, the front panel knob controls volume for both channels

identically. For pan, it controls them using either balance or width, as explained in the previous section for stereo pairs.

Adjusting input trim

When you push the *Mix Volume* knob (Figure 6-4) repeatedly to cycle among the four different mixes, there is a fifth choice: the *trim* LED (Figure 6-4). When it is illuminated, the input volume knobs (Figure 6-1) control the trim level for each input. The trim level adjustment occurs before the signal enters the Audio Express mixer, so it applies to all mixes. Trim is useful for adjusting the overall level of the input. For example, if you plug in a mic, and the signal is so strong (loud) that it distorts the input, use trim to reduce the volume until it no longer overdrives the input.

Toggling pad and 48V phantom power

When adjusting trim (the trim LED is illuminated), you can toggle the -20 dB pad and 48-volt phantom power for each mic input as follows:

- Push the knob to toggle the -20 dB pad
- Push *and hold* the knob to toggle 48V power

When enabled, the corresponding 48V or *pad* LED (Figure 6-4) will illuminate.

Using the Audio Express as a monitor mixer while recording tracks on the computer

You can use the Audio Express front-panel mixer while connected to a computer. This allows you to create monitor mixes that are independent of what’s going on in the computer. Using the Audio Express as a monitor mixer in this fashion is a good way to avoid latency (delay) issues that can arise when using audio software on a computer. See chapter 9, “Reducing Monitoring Latency” (page 45) for more information about dealing with latency.

Using the Audio Express as a stand-alone mixer

When the Audio Express is not connected to a computer, it can operate as a stand-alone mixer, and you can use the front panel to control your mix.

VISUAL FEEDBACK WHEN ADJUSTING MIX CONTROLS

When you turn a knob to adjust volume or pan, the ten-segment MIC/INST ladder LEDs provide visual feedback as you turn the knob. For example, when adjusting pan, the two meters show the relative amount of signal going to the left and right channels.

METERING AND ACTIVITY LEDS

The *MIDI* and *SPDIF* LEDs (Figure 6-2) indicate input and output activity on the MIDI and digital audio jacks.

The MAIN meters (Figure 6-1) provide five-segment metering for the Main Outs.

The LINE meters provide four-segment metering for line inputs 3-4. Below them are signal activity LEDs for line outputs 3-4.

The MIC/INST meters provide ten-segment metering for the mic/instrument combo jacks.

RESTORING FACTORY DEFAULTS

To restore factory default settings hold down the MIX and PHONES buttons simultaneously for three (3) seconds.

CHAPTER 7 Cubase, Nuendo, Live and Other ASIO Software

OVERVIEW

The Audio Express includes an ASIO driver that provides multi-channel I/O and sample-accurate synchronization with Steinberg’s Cubase family of digital audio sequencers, including Cubase and Nuendo.

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WHAT IS ASIO?

ASIO is an acronym for *Audio Streaming Input and Output*. The ASIO MOTU FireWire Audio driver allows the Audio Express to provide multi-channel audio input and output for any audio application that supports ASIO drivers.

Attention: Other software users

The Audio Express ASIO driver also provides multi-channel I/O with any ASIO-compatible audio software. Cubase is used for the examples in this chapter. However, the basic procedures are the same and can be easily applied to any ASIO-compatible software. Just follow the general descriptions at the beginning of each main section in this chapter. Consult your software documentation for details about each topic, if necessary.

If your audio software doesn’t support ASIO

If your host audio software does not support ASIO, but instead supports WDM (or legacy Wave) audio drivers, refer to the next chapter.

PREPARATION

To make sure that everything is ready for Cubase, install Cubase first (if you haven’t already done so), and then see these chapters before proceeding:

- chapter 3, “IMPORTANT! Run the Audio Express Software Installer First” (page 15)
- “Installing the Audio Express Hardware” on page 17.

RUN MOTU AUDIO CONSOLE

Before you run Cubase, launch MOTU Audio Console to configure your Audio Express hardware. MOTU Audio Console lets you configure your audio interface, and it lets you enable the desired inputs and outputs. Only enabled inputs and outputs will be available to Cubase, so this is an important step. For complete details regarding MOTU Audio Console, see chapter 5, “MOTU Audio Console” (page 27).

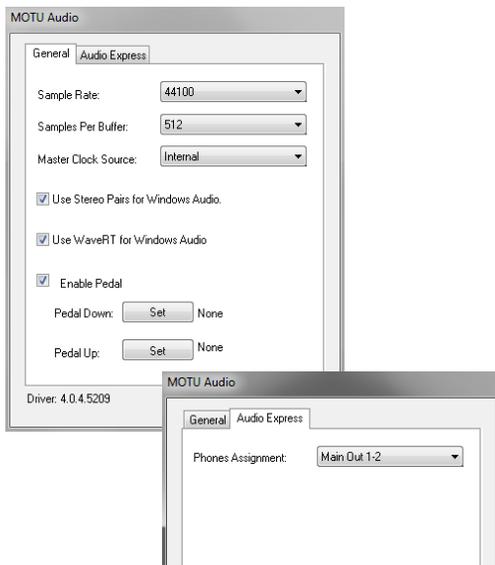


Figure 7-1: MOTU Audio Console gives you access to all of the settings in the Audio Express hardware, including the clock source and sample rate.

For complete details about the Audio Express settings, see chapter 5, “MOTU Audio Console” (page 27). The following sections provide a brief explanation of each Audio Express setting for use with Cubase.

Sample rate

Choose the desired overall sample rate for the Audio Express system and Cubase. Newly recorded audio in Cubase will have this sample rate.

Clock Source

This setting is very important because it determines which audio clock the Audio Express will follow.

If you do not have any digital audio connections to your Audio Express (you are using the analog inputs and outputs only), and you will not be slaving Cubase to external SMPTE time code, choose *Internal*.

If you have a S/PDIF digital audio device connected to the Audio Express, see “Connect and sync S/PDIF devices” on page 22.

If you are slaving the Audio Express and Cubase to SMPTE time code via the Audio Express itself, choose *SMPTE* and follow the directions in “Resolving Cubase or Nuendo to SMPTE time code” on page 77.

☛ If you are using an ASIO host application other than Cubase or Nuendo, it must support the ASIO 2.0 sample-accurate positioning protocol in order to support the Audio Express’s direct SMPTE sync (and sample-accurate sync) feature.

Samples Per Buffer

The *Samples Per Buffer* setting can be used to reduce the delay — or *monitoring latency* — that you hear when live audio is patched through your Audio Express hardware and Cubase. For example, you might have MIDI instruments, samplers, microphones, and so on connected to the analog inputs of the Audio Express. If so, you will often be mixing their live input with audio material recorded in Cubase. See chapter 9, “Reducing Monitoring Latency” (page 45) for complete details.

Phones

This Audio Express setting lets you choose what you’ll hear from the headphone jack. For example, if you choose *Main Out 1-2*, the headphones will duplicate the main outs. Or you can choose any other output pair. If you choose *Phones 1-2*, this setting makes the headphone jack serve as its own independent output pair. As a result, you’ll see *Phones 1-2* as an additional audio destination in Cubase’s audio output menus.

CHOOSING THE MOTU ASIO DRIVER

Once you've made the preparations described so far in this chapter, you're ready to run your audio software and enable the MOTU ASIO Driver. Check the audio system or audio hardware configuration window in your software. There will be a menu there that lets you choose among various ASIO drivers that may be in your system. Choose the MOTU ASIO Driver from this menu.

Cubase and Nuendo

To activate the Audio Express driver in Cubase or Nuendo, go to the Device Setup window, click *VST Audio System* and choose *MOTU Audio ASIO* from the *ASIO Driver* menu (Figure 7-2). Make the other settings in the dialog as need for your system and synchronization scenario.

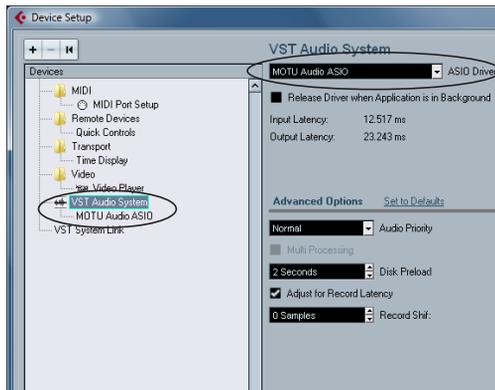


Figure 7-2: Activating the Audio Express ASIO driver in Cubase and Nuendo.

Live

In Ableton Live, access the preferences window and click the *Audio* tab. Choose *ASIO* from the *Driver Type* menu. Choose the *MOTU Audio ASIO* from the *ASIO Driver* menu as shown below in Figure 7-3. To enable or disable Audio Express input or output channels, click the *Input Config* or *Output Config* buttons. To access MOTU Audio Console, click the *Hardware Setup* button.

For information about the *Buffer Size* setting, see “Adjusting the audio I/O buffer” on page 47.

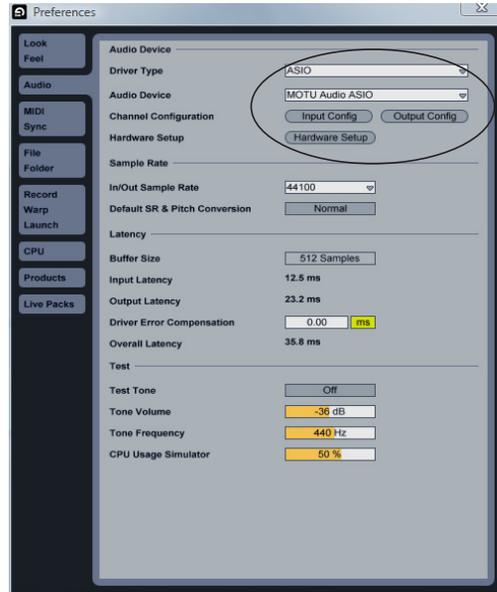


Figure 7-3: Enabling the Audio Express in Live.

Reaper

In Cockos Reaper, access the Preferences and click *Devices* under the *Audio* preferences. Choose *ASIO* for the audio system and choose *MOTU Audio ASIO* from the *ASIO driver* menu.

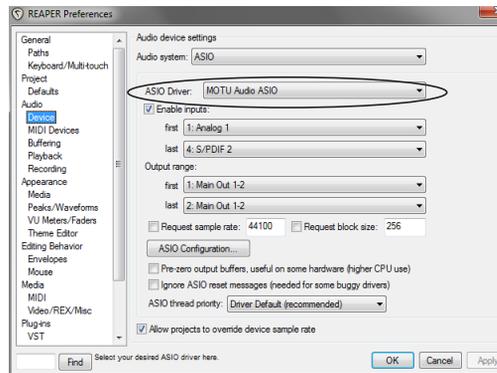


Figure 7-4: Enabling the Audio Express in Reaper.

DIRECT MONITORING

The *Direct Monitoring* option (Figure 7-5) allows you to monitor inputs directly in the Audio Express hardware with no drain on your computer and near zero latency. When you enable this option, Cubase uses the Audio Express’s CueMix FX monitoring features whenever you use Cubase’s monitoring features. For further information, see “Controlling CueMix from within Cubase or Nuendo” on page 49.

OTHER SYSTEM DIALOG SETTINGS

Consult your Cubase or Nuendo documentation for details about the rest of the settings in this dialog.

MANAGING AUDIO EXPRESS INPUTS AND OUTPUTS

Once you’ve chosen the MOTU ASIO Driver in Device Setup as explained earlier in “Choosing the MOTU ASIO driver” on page 37, click *MOTU Audio ASIO* in the left-hand list to see the Audio Express inputs and outputs in the port list on the right (Figure 7-5). To activate an Audio Express output, create a bus in the VST Connections window and assign the desired output to it.

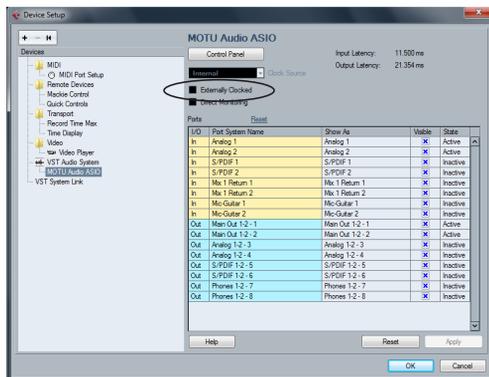


Figure 7-5: Creating Audio Express inputs in Nuendo or Cubase.

Mix1 1-2

In the VST Inputs list, you’ll see an Audio Express input pair called *Mix1 1-2*. This is a stereo feed from the Audio Express that matches its main outs. This can be used, for example, to record a final stereo mix for reference and archiving purposes.

⚠ Warning: the Mix1 inputs can cause feedback loops! DO NOT assign this input to a track assigned to the Audio Express main outs.

For Mix1 1-2, you can choose whether or not to include audio being sent to the main outs from the computer. See “Mix1 Return Includes Computer Output” on page 56.

The “Phones 1-2” output

Host audio tracks assigned to the *Phones 1-2* output pair will be heard on the headphone jack only. If the Phones are set to mirror another output (such as the main outs), the mirrored audio stream is mixed with the phones audio output. For further explanation, see “Phones Assign” on page 30.

CHANGING AUDIO EXPRESS SETTINGS

To change the Audio Express settings at any time, go to the Device Setup window in Nuendo or Cubase and click the Control Panel button, as shown in Figure 7-5. Be sure to click the Reset button after making any changes.

PROCESSING LIVE INPUTS WITH HOST-BASED VST PLUG-INS

If you patch a live input (such as MIDI synthesizer) through a VST plug-in effect in Cubase, you might hear a slight delay. There are several ways to reduce this delay. For details, see chapter 9, “Reducing Monitoring Latency” (page 45).

WORKING WITH CUEMIX FX MIXING

The Audio Express provides its own hardware-based mixing, which you can operate hand-in-hand with your host software’s complete mixing environment. For example, the Audio Express can serve as a monitor mixer routing

channels to musicians, or it can serve as an integrated extension of your Cubase/Nuendo mixing environment. If you program an Audio Express mixing configuration that goes hand in hand with your Cubase/Nuendo project, be sure to use the file save features in CueMix FX to save the Audio Express settings as a file in your Cubase/Nuendo project folder for instant recall of all settings. See chapter 10, “CueMix FX” (page 51) for complete details.

SYNCHRONIZATION

Together, Cubase (or Nuendo) and the Audio Express have the ability to resolve directly to SMPTE time code, without an additional time code synchronizer. See “Syncing to SMPTE time code” on page 78.

MIDI I/O

Once you’ve run the Audio Express software installer as explained in “Installing the Audio Express software” on page 15, the Audio Express MIDI ports will appear as a MIDI input source and output destination in your host software’s MIDI input and output menus.

24-BIT OPERATION

Your Audio Express hardware fully supports Cubase and Nuendo’s 24-bit recording capabilities. Simply enable 24-bit operation as instructed in your Cubase or Nuendo manual. The Audio Express always supplies a 24-bit data stream, and when you enable 24-bit operation in Cubase or Nuendo, it simply uses all 24-bits supplied by the Audio Express hardware.

CHAPTER 8 SONAR and other WDM Software

OVERVIEW

The Audio Express WDM driver provides standard multi-channel input and output for WDM-compatible audio software running under Windows.

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WDM AND WAVE DRIVER COMPATIBILITY

WDM is an acronym for *Windows Driver Model*. The MOTU WDM Driver allows the Audio Express to provide multi-channel audio input and output for any audio application that supports WDM audio drivers.

Attention: Cakewalk users

Cakewalk SONAR is used for the WDM driver setup examples in this chapter.

Attention: other software users

SONAR is used for the WDM driver setup examples in this chapter. However, the basic procedures are the same and can be easily applied to any WDM-compatible software. Consult your software documentation for details, if necessary.

INSTALLING THE MOTU AUDIO WDM DRIVER

The MOTU Audio Installer CD installs the MOTU WDM driver into Windows for you. That's pretty much all the preparation you need. See chapter 3, "IMPORTANT! Run the Audio Express Software Installer First" (page 15).

WaveRT SUPPORT

The Audio Express Windows driver also supports WaveRT, a low-latency audio driver standard developed by Microsoft. WaveRT is supported by Sonar 8 and later, under Windows 7 and Vista. To enable WaveRT, check the *Use WaveRT for Windows Audio* option in MOTU Audio Setup.

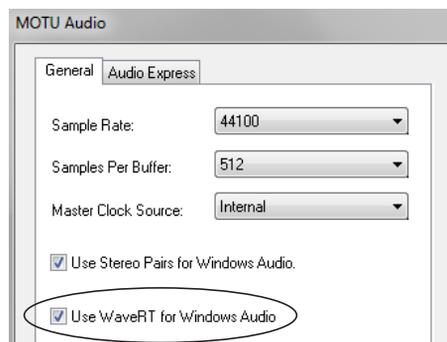


Figure 8-1: The WaveRT option can be enabled in the MOTU Audio Setup console.

If you uncheck this box, WaveRT support is disabled, and legacy WDM driver support is provided instead.

MAKING SETTINGS IN MOTU AUDIO CONSOLE

Before you run your audio software, launch MOTU Audio Console to configure your Audio Express hardware. MOTU Audio Console lets you choose the audio clock source, sample rate, buffer size and other hardware settings. For details, see chapter 5,

“MOTU Audio Console” (page 27). The following sections provide a brief explanation of each Audio Express setting for use with SONAR.

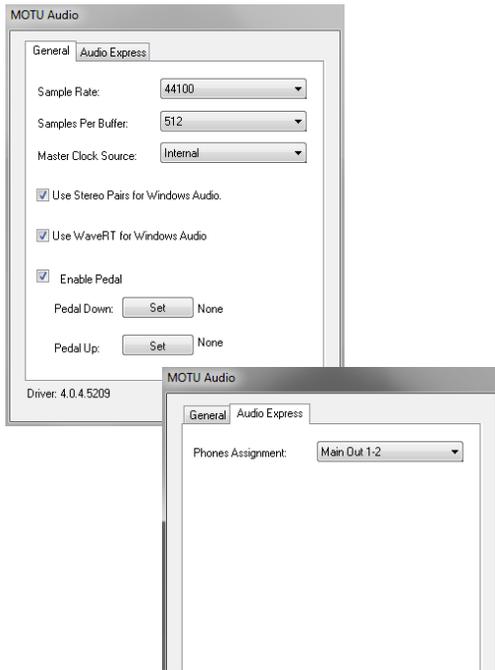


Figure 8-2: MOTU Audio Console gives you access to all of the settings in the Audio Express hardware, including the clock source, sample rate and headphone output assignment.

Sample rate

Choose the desired overall sample rate for the Audio Express system and SONAR. Newly recorded audio in SONAR will have this sample rate.

Clock Source

This setting is very important because it determines which audio clock the Audio Express will follow.

If you do not have any digital audio connections to your Audio Express (you are using the analog inputs and outputs only), and you will not be slaving SONAR to external SMPTE time code, choose *Internal*.

If you have a S/PDIF digital audio device connected to the Audio Express, see “Connect and sync S/PDIF devices” on page 22.

Samples Per Buffer

The *Samples Per Buffer* setting can be used to reduce the delay — or *monitoring latency* — that you hear when live audio is patched through your Audio Express hardware and SONAR. For example, you might have MIDI instruments, samplers, microphones, and so on connected to the analog inputs of the Audio Express. If so, you will often be mixing their live input with audio material recorded in SONAR. See chapter 9, “Reducing Monitoring Latency” (page 45) for complete details.

Phones

This Audio Express setting lets you choose what you’ll hear from the headphone jack. For example, if you choose *Main Out 1-2*, the headphones will duplicate the main outs. Or you can choose any other output pair. If you choose *Phones 1-2*, this setting makes the headphone jack serve as its own independent output pair. As a result, you’ll see *Phones 1-2* as an additional audio destination in SONAR’s audio output menus.

ENABLING THE MOTU AUDIO WDM DRIVER

Once you’ve made the preparations described so far in this chapter, you’re ready to run your audio software and enable the MOTU Audio WDM driver. Check the audio system or audio hardware configuration window in your software.

Enabling the MOTU Audio WDM driver in SONAR

To activate the MOTU Audio WDM driver in SONAR:

- 1 Go to Preferences.
- 2 In the Audio preferences section, choose *Devices*.

3 Check the Audio Express inputs and outputs that you wish to use and uncheck the ones you don't as shown in Figure 8-3.

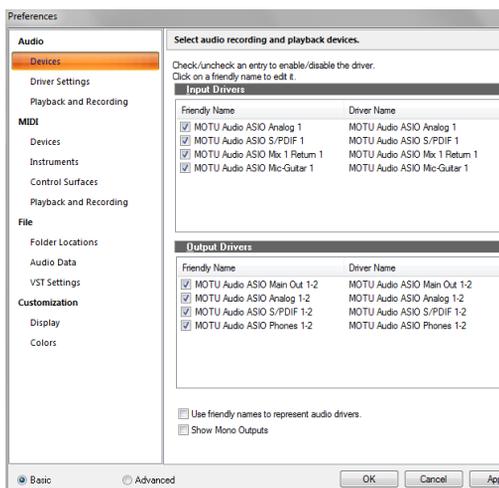


Figure 8-3: Activating the MOTU Audio WDM driver in SONAR by highlighting Audio Express inputs and outputs.

Enabling the MOTU ASIO driver in SONAR

As an alternative to the Audio Express's WDM driver, you can use the Audio Express's ASIO driver. Note, however, that you can only use one audio interface at a time under ASIO. In the Audio preferences section, choose *Playback and Recording* and choose *ASIO* from the *Driver Mode* menu, as shown below:

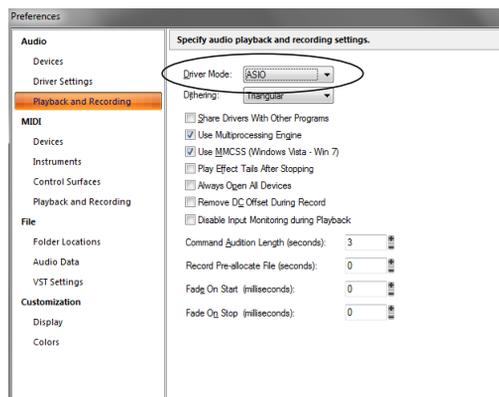


Figure 8-4: Enabling the ASIO driver instead of the WDM driver.

Other audio settings in SONAR

There are a few other settings in SONAR that impact Audio Express operation:

- 1 In the Audio preferences section, choose *Driver Settings*.
- 2 Choose any Audio Express input and output for recording and playback timing. It doesn't matter which ones.
- 3 In the same settings, click the Wave Profiler button and run the Wave Profiler.

This process chooses the optimum settings for the Audio Express hardware.

WORKING WITH AUDIO EXPRESS INPUTS AND OUTPUTS

Once you've enabled the Audio Express inputs and output, Audio Express audio inputs and outputs will appear in your host software's input/output menus, and you can set them up and use them as any standard audio input and output.

Mix1 1-2

In SONAR, you'll see an Audio Express input pair called *Mix1 1-2*. This is a stereo feed from the Audio Express that matches its main outs. This can be used, for example, to record a final stereo mix for reference and archiving purposes.

 **Warning:** the Return inputs can cause feedback loops! DO NOT assign this input to a track assigned to the Audio Express main outs.

For *Mix1 1-2*, you can choose whether or not to include audio being sent to the main outs from the computer. See "Mix1 Return Includes Computer Output" on page 56.

The "Phones 1-2" output

Host audio tracks assigned to the *Phones 1-2* output pair will be heard on the headphone jack only. If the Phones are set to mirror another output

(such as the main outs), the mirrored audio stream is mixed with the phones audio output. For further explanation, see “Phones Assign” on page 30.

CHANGING AUDIO EXPRESS SETTINGS

You can change the Audio Express settings at any time by accessing MOTU Audio Console.

REDUCING DELAY WHEN MONITORING LIVE INPUTS

If you have live audio inputs connected to the Audio Express, such as MIDI synthesizers, samplers, microphones or other live instruments, you might hear a slight delay when their audio is being monitored through your Audio Express hardware and your host audio program. There are several ways to reduce — and eliminate — this audible monitoring delay. For details, see chapter 9, “Reducing Monitoring Latency” (page 45).

PROCESSING LIVE INPUTS WITH HOST-BASED VST PLUG-INS

If you patch a live input (such as MIDI synthesizer) through a plug-in effect in SONAR, you might hear a slight delay. There are several ways to reduce this delay. For details, see chapter 9, “Reducing Monitoring Latency” (page 45).

WORKING WITH CUEMIX FX MIXING

The Audio Express provides its own hardware-based mixing, which you can operate hand-in-hand with your host software’s complete mixing environment. For example, the Audio Express can serve as a monitor mixer routing channels to musicians, or it can serve as an integrated extension of your SONAR mixing environment. If you program an Audio Express mixing configuration that goes hand in hand with your SONAR project, be sure to use the file save features in CueMix FX to save the Audio Express settings as a file in your SONAR project folder for instant recall of all settings. See chapter 10, “CueMix FX” (page 51) for complete details.

MIDI I/O

Once you’ve run the Audio Express software installer as explained in “Installing the Audio Express software” on page 15, the Audio Express MIDI ports will appear as a MIDI input source and output destination in your WDM-compatible software’s MIDI input and output menus.

24-BIT OPERATION

Your Audio Express hardware fully supports 24-bit recording in any audio software that supports it. Simply enable 24-bit operation as instructed by the software. The Audio Express system always supplies the software with a 24-bit data stream, and when you enable 24-bit operation, it simply uses all 24-bits supplied by the Audio Express hardware.

CHAPTER 9 Reducing Monitoring Latency

OVERVIEW

Monitoring latency is that slight delay you hear when you run an input signal through your host audio software. For example, you might hear it when you drive a live mic input signal through an amp modeling plug-in running in your audio sequencer.

This delay is caused by the amount of time it takes for audio to make the entire round trip through your computer, from when it first enters an Audio Express input, passes through the Audio Express hardware into the computer, through your host audio software, and then back out to an Audio Express output.

If you don't need to process a live input with plug-ins, the easiest way to avoid monitoring latency is to use the Audio Express's CueMix digital mixer to patch the input directly to your monitor outs via the Audio Express audio hardware. For details, see "CueMix hardware monitoring" on page 48.

If you *do* need to process a live input with plug-ins, or if you are playing virtual instruments live through your Audio Express audio hardware, you can significantly reduce latency — and even make it completely inaudible, regardless of what host audio application software you use. This chapter explains how.

It is important to note that monitoring delay has no effect on when audio data is recorded to disk or played back from disk. Actual recording and playback is extremely precise.

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MONITORING LIVE INPUT

There are two ways to monitor live audio input with an Audio Express: 1) through the computer or 2) via the Audio Express's CueMix hardware mixer. Figure 9-1 below shows method 1, which allows you to add effects processing such as reverb and guitar amp effects via plug-ins in your audio software. See the next section, “Adjusting the audio I/O buffer” for details about how to reduce — and possibly eliminate — the audible monitoring delay that the computer introduces.

Figure 9-2 shows how to use CueMix hardware-based monitoring, which lets you hear what you are recording with no monitoring delay and no computer-based effects processing. Instead, input

is routed directly to an output. See “CueMix hardware monitoring” later in this chapter for details on how to use CueMix with your audio software, or with the included CueMix FX software.

If the material you are recording is suitable, there is a third way to monitor live input: use both methods (Figure 9-1 and Figure 9-2) at the same time. For example, you could route guitar to both the computer (for an amp model effect) and mix that processed signal on the main outs with dry guitar from CueMix.

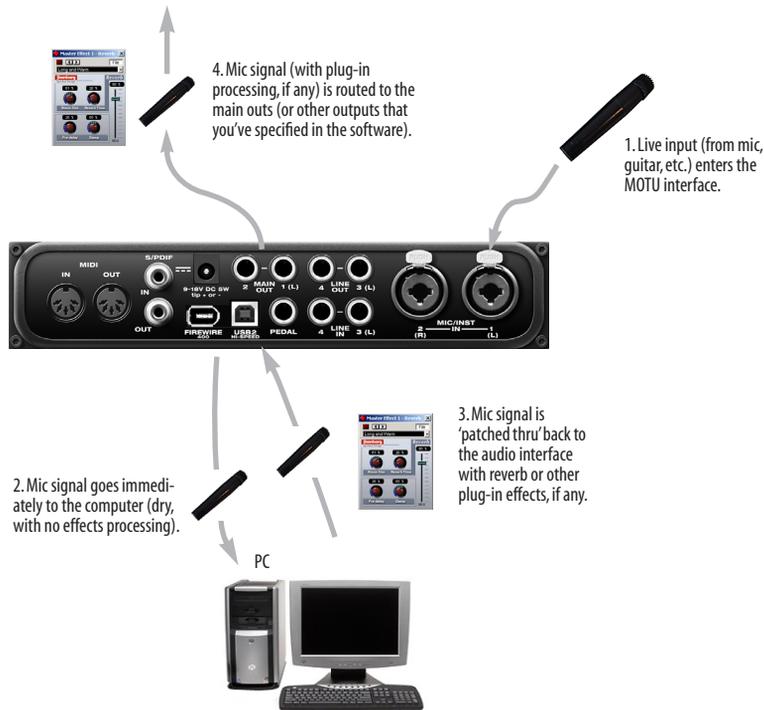


Figure 9-1: There are two ways to monitor live audio inputs with an Audio Express: 1) through the computer or 2) via CueMix hardware monitoring. This diagram shows method 1 (through the computer). When using this method, use your host software's buffer setting to reduce the slight delay you hear when monitoring the live input, but don't lower it too much, or your computer might get sluggish.

ADJUSTING THE AUDIO I/O BUFFER

A *buffer* is a small amount of computer memory used to hold data. For audio interfaces like the Audio Express, buffers are used for the process of transferring audio data in and out of the computer. The size of the buffers determines how much delay you hear when monitoring live inputs through your audio software: larger buffers produce more delay; smaller buffers produce less.

Buffer size adjustment is made in MOTU Audio Console, as shown in Figure 9-3 via the *Samples Per Buffer* setting.

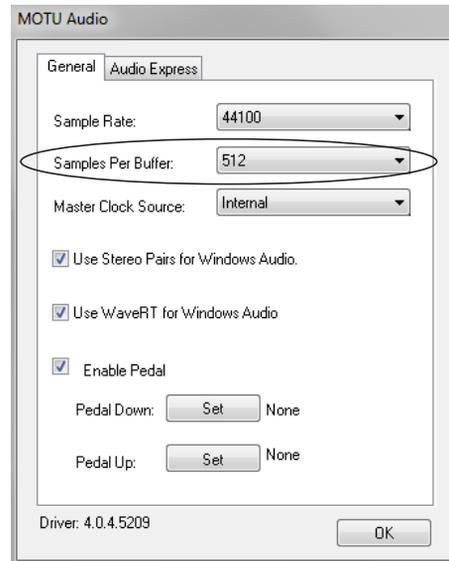


Figure 9-3: Lowering the 'Samples Per Buffer' setting in MOTU Audio Console reduces patch thru latency. But doing so increases the processing load on your computer, so keep an eye on the Performance Monitor in your host audio software.

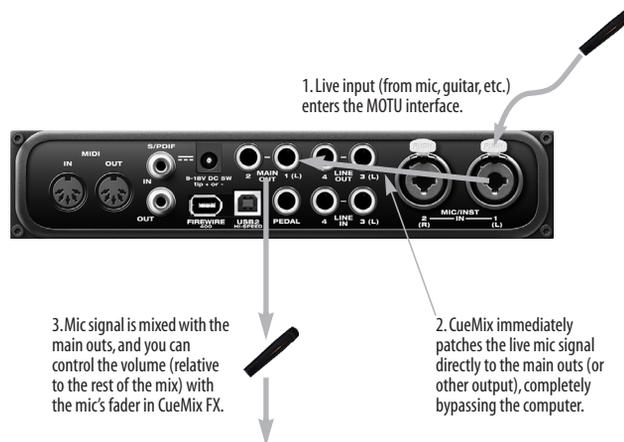


Figure 9-2: This diagram shows the signal flow when using CueMix no-latency monitoring. Notice that this method does not process the live input with plug-ins in your audio software.

Lower latency versus higher CPU overhead

The buffer setting has a large impact on the following things:

- Patch thru latency
- The load on your computer's CPU
- Possible distortion at the smallest settings
- How responsive the transport controls are in your audio software

The buffer setting presents you with a trade-off between the processing power of your computer and the delay of live audio as it is being patched through your software. If you reduce the size, you reduce patch thru latency, but significantly increase the overall processing load on your computer, leaving less CPU bandwidth for things like real-time effects processing. On the other hand, if you increase the buffer size, you reduce the load on your computer, freeing up bandwidth for effects, mixing and other real-time operations.

If you are at a point in your recording project where you are not currently working with live, patched-thru material (e.g. you're not recording vocals), or if you have a way of externally processing inputs, choose a higher buffer size. Depending on your computer's CPU speed, you might find that settings in the middle work best (256 to 1024).

Transport responsiveness

Buffer size also impacts how quickly your audio software will respond when you begin playback, although not by amounts that are very noticeable. Lowering the buffer size will make your software respond faster; raising the buffer size will make it a little bit slower, but barely enough to notice.

Effects processing and automated mixing

Reducing latency with the buffer size setting has another benefit: it lets you route live inputs through the real-time effects processing and mix automation of your audio software.

CUEMIX HARDWARE MONITORING

The Audio Express has a more direct method of patching audio through the system: *CueMix*. CueMix is a digital mixer in the Audio Express hardware itself. CueMix has two important benefits:

- First, it completely eliminates the patch thru delay (reducing it to a small number of samples — the same amount as any digital mixer).
- Secondly, CueMix FX imposes no strain on the computer.

The trade-off, however, is that CueMix FX bypasses your host audio software. Instead, live audio inputs are patched directly through to outputs in the Audio Express itself and are mixed with disk tracks playing back from your audio software. This means that you cannot apply plug-ins, mix automation, or other real-time effects that your audio software provides. But for inputs that don't need these types of features, CueMix is the way to go.

On the other hand, if you really need to use the mixing and processing provided by your audio software, you should not use CueMix. Instead, reduce latency with the buffer setting (as explained earlier in this chapter).

THREE METHODS FOR CONTROLLING CUEMIX

There are three ways to control CueMix:

- With the knobs on the Audio Express front panel
- With the Audio Express CueMix FX software
- From within your host audio software (if it supports direct hardware monitoring)

You can even use these methods simultaneously.

Controlling CueMix from the front panel

The knobs on the Audio Express front panel give you direct control over CueMix. For complete details, see “Mixing with the front panel controls” on page 32.

Using the CueMix FX application

If your host audio software does not support direct hardware monitoring, you run the Audio Express’s CueMix FX application side-by-side with your host software and manage your monitor mix in CueMix FX. For complete details, see chapter 10, “CueMix FX” (page 51).

Controlling CueMix from your audio software

Some ASIO-compatible audio applications, such as Cubase and Nuendo, allow you to control CueMix monitoring from within the application (without the need to use the separate CueMix FX software). In most cases, this support consists of patching an Audio Express input directly to an output when you record-arm a track. Exactly how this is handled depends on the application.

The following applications are among those that support direct control over CueMix:

- Cubase
- Nuendo

CueMix routings that are made via host applications are made “under the hood”, which means that you won’t see them in CueMix FX. However, CueMix connections made inside your host audio software dovetail with any other mixes you’ve set up in CueMix FX. For example, if your host application routes audio to an output pair that is already being used in CueMix FX, both audio streams will simply be merged to the output.

Controlling CueMix from within Cubase or Nuendo

To turn on CueMix in Cubase or Nuendo, enable the Direct Monitoring check box in the Device Setup window (Figure 7-2 on page 37).

Other ASIO 2.0-compatible host software

If your ASIO-compatible host audio software supports ASIO’s direct monitoring feature, consult your software documentation to learn how to enable this feature. Once enabled, it should work similarly as described for Cubase (as explained in the previous section).

Using CueMix with WDM- or Wave-compatible software

Run CueMix FX (chapter 10, “CueMix FX” (page 51)) and use it to route live inputs directly to outputs, control their volume and panning, etc. If necessary, you can save your CueMix FX setup with your project file.

CHAPTER 10 CueMix FX

OVERVIEW

CueMix FX is a cross-platform software application that provides graphic, on-screen control for the Audio Express’s mixing features. Essentially, it gives you on-screen control of the Audio Express’s front panel mixing controls.

CueMix FX also provides many advanced audio analysis tools, including a tuner and oscilloscope.

CueMix FX can be used independently of host audio software, or together with it. CueMix dovetails with the direct monitoring features of your host audio software, allowing you to seamlessly mix in both environments.

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ADVANTAGES OVER HOST-BASED MIXING AND PROCESSING

CueMix FX provides several advantages over mixing and processing in your host audio software:

- CueMix has no buffer latency. Thanks to the Audio Express’s DSP chip, CueMix provides the same throughput performance as a digital mixer.
- CueMix routing can be maintained independently of individual software applications or projects.
- CueMix routing can operate without the computer, allowing the Audio Express to operate as a portable, stand-alone mixer.

CUEMIX FX INSTALLATION

CueMix FX is installed as a standard application by the Audio Express setup software.

CUEMIX FX BASIC OPERATION

The CueMix FX application is simple to operate, once you know these basic concepts.

A separate mix for each output pair

As explained in “Mixing with the front panel controls” on page 32, the Audio Express provides a separate mix for each of its four output pairs, as follows:

Mix name	Physical output
Mix 1	Main Out 1-2
Mix 2	Analog 1-2
Mix 3	S/PDIF 1-2
Mix 4	Phones 1-2

Many inputs to one output pair

It might be useful to think of each mix as some number of inputs all mixed down to a stereo output pair. CueMix FX lets you choose which inputs to include in the mix, and it lets you specify the level and pan for each input being fed into the mix.

Viewing one mix at a time

CueMix FX displays one mix at a time. To select which mix you are viewing, choose it from the mix menu (Figure 10-1). Click the mix name to change it. The mix output pair appears above the mix master fader (Figure 10-1).

Each mix is independent

Each mix has its own settings. Settings for one mix will not affect another. For example, if an input is used for one mix, it will still be available for other mixes. In addition, inputs can have a different volume, pan, mute and solo setting for each mix.

Widening the CueMix FX window

To view more input faders at once, drag the right-hand edge of the window to the right.

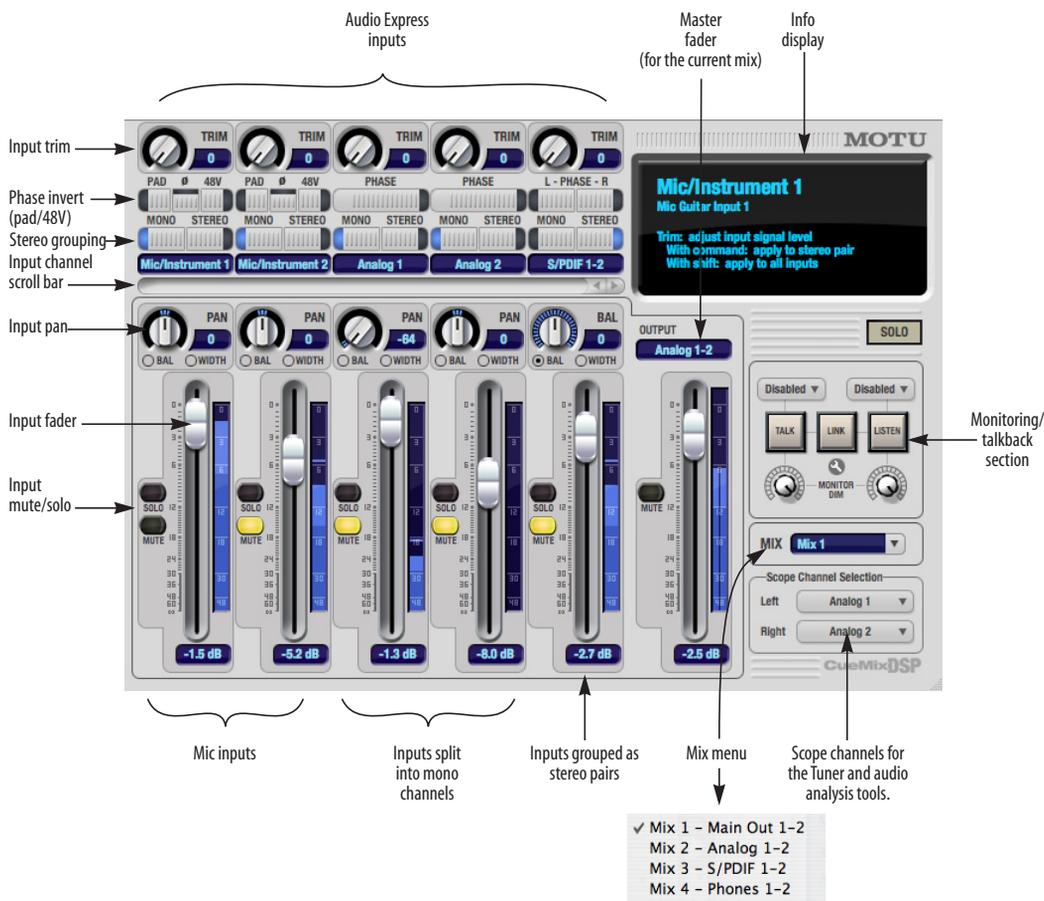


Figure 10-1: CueMix FX is a virtual mixer that gives you control over the Audio Express's on-board mixing features.

WORKING WITH A MIX

Each mix has the following components:

- A stereo output with master fader
- Name
- Master mute (to enable/disable the entire mix)
- Any number of mono or stereo inputs
- Pan, volume, mute and solo for each input

Viewing a mix

To view a mix, choose it from the *Mix* menu (Figure 10-1). The menu shows all mixes by name, followed by the audio interface's output pair that it corresponds to.

Naming a mix

Click the mix name to edit its text.

Master fader

The master fader (Figure 10-1) controls the overall level of the mix (its volume on its stereo output). Use the individual input faders to the left to control individual input levels.

Mix mute

The mix mute button (Figure 10-1) disables (silences) the mix.

Mix level meter

The mix level meter, which is post-fader, shows you the output for the mix's output.

Input channels

The *Inputs* section (Figure 10-1) gives you access to individual Audio Express inputs (or input pairs). Unmute the input and bring up its fader to hear the input on the current mix's output. Mute, solo, and pan are also provided for each input channel.

Input trim

All Audio Express inputs, both analog and digital, offer continuously variable input trim. In all cases, trim level can be controlled digitally in

approximately 1 dB increments. This includes the digitally controlled analog trims on the two mic/guitar inputs and the two quarter-inch analog inputs on the back panel. Here is a summary of input trim ranges for each type of Audio Express input:

Input	Trim cut	Trim boost	Trim Range
Mic/Guitar	0 dB	+60 dB	60 dB
TRS analog inputs	0 dB	+22 dB	22 dB
S/PDIF	0 dB	+12 dB	12 dB

Once you adjust the trim levels, you can save them as a file on disk for future instant recall. See “Saving and loading hardware presets” on page 56 and “Configurations menu” on page 72.

Pad and phantom

The Pad and phantom settings are the same 20 dB pad and 48V phantom power settings that you can also control from the Audio Express front panel encoders. There are separate settings for each mic input.

Invert phase

The *Phase* button (Figure 10-1) inverts the phase of the input signal. For stereo pairs, you can invert the phase for the left and right channels independently. For the mic/instrument channels, click the phase button (\emptyset) repeatedly to cycle among two states for mono channels (on or off) and four states for stereo grouped channels (no inversion, left only, both, or right only).

Mono/stereo pairing

Click the *Mono* button (Figure 10-1) if you would like an input to be treated as a mono channel. If you would like to work with it as one channel of a linked stereo pair, click the *Stereo* button. Inputs are grouped in odd/even pairs (Mic/instrument 1-2, Analog 1-2, etc.) Stereo pairs appear as a single channel strip in the CueMix FX mixer.

Input pan

The pan knob (Figure 10-1) for each input pans the input across the mix's stereo outputs. If the input itself is grouped as a stereo pair, two forms of panning control are provided:

Balance

Balance works like the balance knob on some radios: turn it left and the right channel dims, turn it right and left channel dims. But the left channel always stays left and the right channel stays right.

Width

Width spreads the left and right channels across the stereo image, depending on the knob position. Maximum value (turning the pan knob all the way up) maintains the original stereo image: the left channel goes entirely left and right goes entirely right, without attenuation. The minimum value (turning the knob all the way down) creates a mono effect: equal amounts of left and right are combined and sent to both outputs. In between, the left out is a mixture of the left input and some of the right input (and vice-versa) with the effect of narrowing the field.

Input fader

Use the input fader (Figure 10-1) to adjust the level for the input in the mix. Note that an input can have different level, pan, mute and solo settings for each mix. Input channel level meters are post-fader.

SHORTCUTS

Use these general modifier keys as shortcuts:

Shortcut	Result
Shift key	Applies your action to all inputs or all outputs in the mix.
Control key	Applies your action to the stereo input pair, even when it is currently configured as mono.
Alt key	Applies your action to all busses.
Shift-Alt	Applies your action to all inputs and mixes.
Double-click	Returns the control to its default value (pan center, unity gain, etc.)

INFO DISPLAY

The info display shows fly-over help for items in the CueMix FX window.

TALKBACK AND LISTENBACK

CueMix FX provides *Talkback* and *Listenback* buttons. Talkback allows an engineer in the control room to temporarily dim all audio and talk to musicians in the live room. Conversely, Listenback allows musicians to talk to the control room.

Hardware setup

Figure 10-2 below shows a typical hardware setup for Talkback and Listenback. For Talkback, set up a dedicated mic in your control room and connect it to a mic input on your Audio Express. For Listenback, set up a dedicated listenback mic in the live room for the musicians and connect it to another mic input. For talkback output, set up a headphone distribution amp or set of speakers in the live room, and connect it to a pair of outputs. For example, you might connect a headphone distribution amp to analog outputs 3-4 on your Audio Express, as demonstrated below in Figure 10-2.

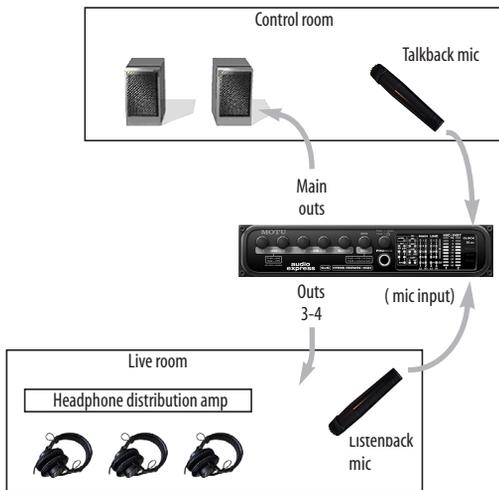


Figure 10-2: Typical hardware setup for Talkback and Listenback.

Talkback / Listenback mic input

Choose the audio input to which your Talkback and/or Listenback mic is connected, as shown below:

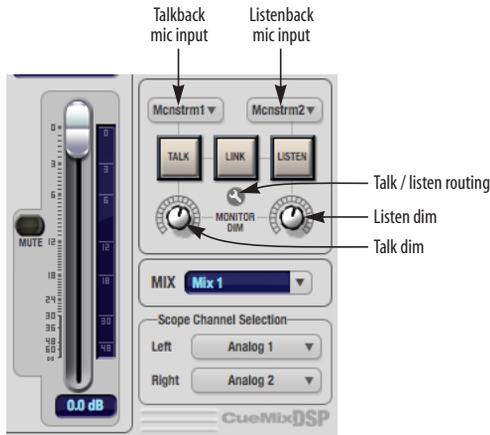


Figure 10-3: Specifying the talkback and listenback mic inputs. Use the input fader for the chosen input to control the mic volume.

Talkback / Listenback monitor dim

Use the knobs next to the Talk and Listen buttons (Figure 10-3) to determine the amount of attenuation you would like to apply to all other audio signals (besides the talkback/listenback signal) when Talkback and/or Listenback is

engaged. To completely silence all other CueMix audio, turn them all the way down. Attenuation only occurs when talkback or listenback is engaged. Audio playing back from disk (your host software) is not affected.

Talk / Listen signal routing

Click the *Talk/Listen routing* button (Figure 10-3) to open the routing dialog (Figure 10-4). Check the boxes next to the outputs on which you'd like to hear the Talkback mic and/or Listenback mic. For example, as demonstrated in the diagram in Figure 10-2, to hear the Talkback mic on the headphones in the live room, check the *Analog 1-2* check box in the *Talk* column as demonstrated in Figure 10-4. To hear the Listenback mic on the main monitors in the control room, check the *Main Out 1-2* check box in the *Listen* column.

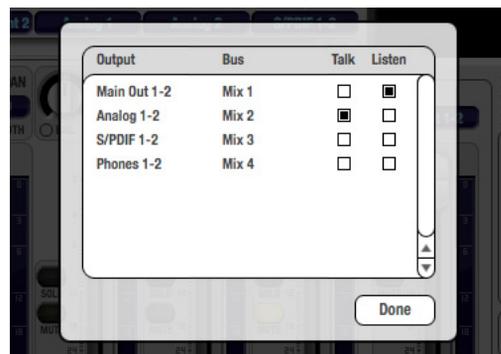


Figure 10-4: Routing the Talkback and Listenback mics to outputs.

Engaging/disengaging Talkback and Listenback

To engage Talk back or Listenback, press on the Talk or Listen buttons (Figure 10-3) and then release to disengage. Alt-click to make the buttons “sticky” (stay engaged until you click them again — so you don’t have to hold down the mouse). Or use the Talkback menu items.

If you would like to engage both Talkback and Listenback at the same time, enable the *Link* button (Figure 10-3).

Controlling Talkback and Listenback volume

To control the volume of the Talkback and/or Listenback mics, adjust their input fader in CueMix FX. This fader controls the volume of the input, regardless of which bus mix is being displayed in the CueMix FX window. In other words, once an input has been designated as a Talkback or Listenback input, its fader becomes global for all CueMix buses.

SCOPE CHANNEL SELECTION

The *Scope channel selection* menus (Figure 10-1 on page 52) let you choose what input channels will be fed to the CueMix FX audio analysis (scoping) features. See “Devices menu” on page 57.

FILE MENU

Saving and loading hardware presets

The Audio Express can store up to 16 presets in its on-board memory. A preset includes all CueMix FX settings for all for mix busses, but it excludes global settings like clock source and sample rate.

The Load Hardware Preset and Save Hardware Preset commands in the CueMix FX file menu let you name, save and load presets in the Audio Express.

Peak/hold time

In CueMix FX, a peak indicator is a line (representing a virtual LED) displayed in a level meter that indicates the maximum signal level registered by the meter. The *Peak/hold time* setting (File menu) determines how long this indicator remains visible before it disappears (or begins to drop). To disable peak/hold indicators altogether, choose Off from this sub-menu.

Mix1 Return Includes Computer Output

When checked (enabled), the *Mix1 return includes computer Output* File menu item causes audio being sent to the main outs (Mix 1) from host software on the computer to be included in the

Mix1 1-2 return bus. This feature can be used to apply the audio analysis tools, such as the FFT display and oscilloscope, to computer audio output. See “Scoping host software audio output” on page 57.

 **Warning!** When this option is enabled, it can cause feedback loops! In your host software, make sure that the stereo return bus is not routed back to the main outs!

For more information on the Mix 1 Return feature, see the section below that applies to you:

- Cubase/VST hosts — “Mix1 1-2” on page 38
- SONAR/WDM hosts — “Mix1 1-2” on page 43

Hardware follows console stereo settings

This File menu item applies to other MOTU interfaces products and has no effect on the Audio Express.

EDIT MENU

Undo/Redo

CueMix FX supports multiple undo/redo. This allows you to step backwards and forwards through your actions in the software.

Copying & pasting (duplicating) entire mixes

To copy and paste the settings from one mix to another:

- 1 Select the source mix (Figure 10-1) and choose Copy from the Edit menu (or press control-C).
- 2 Choose the destination mix and choose Paste from the Edit menu (or press control-V).

Clear Peaks

Choose *Clear Peaks* from the Edit menu to clear all peak indicators in all CueMix FX meters.

DEVICES MENU

If you are working with more than one MOTU audio interface product, this menu displays all interfaces that are currently on line. Choose any device from the menu to edit its settings using the CueMix FX software.

Below each device are five signal analysis tools: FFT Analysis, Oscilloscope, X-Y Plot, Phase Analysis, and the Tuner. Choose one to open its window. For details on these features, see the following sections.

FFT Analysis / spectrogram	57
Oscilloscope	59
X-Y Plot	65
Phase Analysis	68
Tuner	72

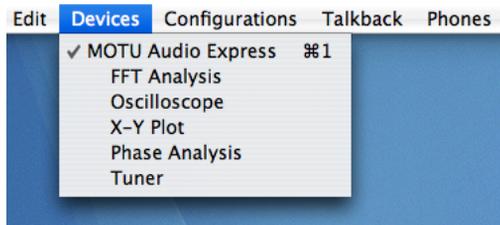


Figure 10-6: Opening the FFT, oscilloscope, X-Y Plot, Phase Analysis, and Tuner windows.

Scope channels

The *Scope channel selection* menus (Figure 10-1 on page 52) let you choose what input channels will be fed to the CueMix FX audio analysis (scoping) features described in the following sections.

Scoping the main outs

If you wish to scope audio output, send it to the Audio Express main outs (Mix 1) and choose one or both of the Mix1 returns from the Scope Channel Selection menus.

Scoping host software audio output

If you want to scope audio output from your host software, send your host's output to the main outs (Mix 1), enable the *Mix1 Return Includes Computer Output* option (File menu), and choose one or both of the Mix1 returns from the Scope Channel Selection menus.

FFT ANALYSIS / SPECTROGRAM

Choose *FFT Analysis* from the Devices menu (Figure 10-6) to open a new window that displays a real-time Fast Fourier Transform (FFT) frequency measurement curve, as shown in Figure 10-5. Each MOTU audio interface has its own FFT/spectrogram window.

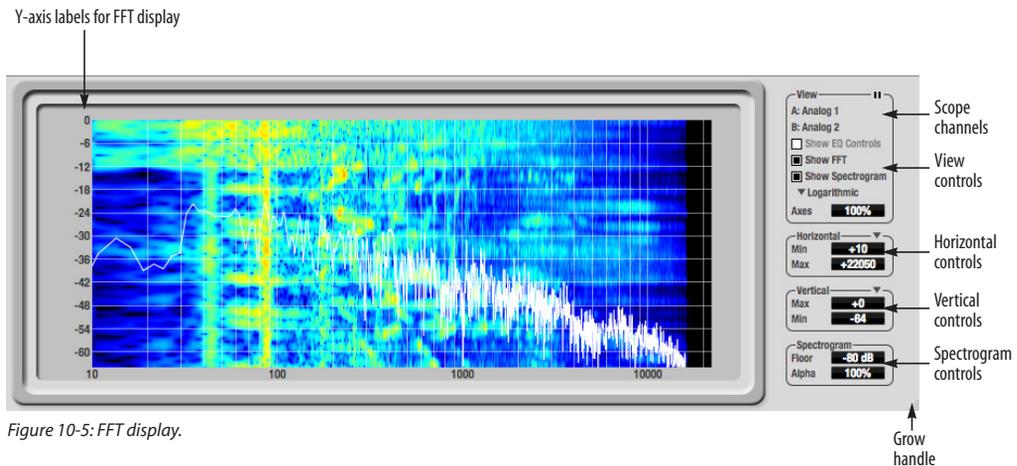


Figure 10-5: FFT display.

Choosing a channel to display

The FFT window follows the currently chosen *Scope channels* (Figure 10-1 on page 52).

View controls

You can show and hide the FFT display and spectrogram as desired using the *View* controls (Figure 10-7).

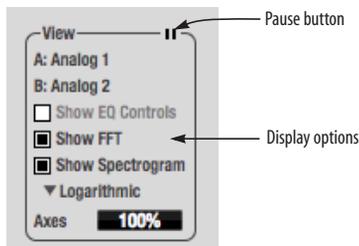


Figure 10-7: View controls

The *Show EQ controls* item is for other MOTU audio interfaces and is grayed out for the Audio Express.

Spectrogram

Click the *Show Spectrogram* option (Figure 10-7) to superimpose a real-time spectrogram “waterfall” display in the background of the FFT display, as demonstrated in Figure 10-5.

The spectrogram scrolls from top to bottom, where the top edge of the display represents what you are hearing “now”. Color represents amplitude along the left/right frequency spectrum. The amplitude color scale runs from black (silence) to red (full scale) as follows:

Silence —————> Full scale
Black —> Blue —> Green —> Yellow —> Orange —> Red

Figure 10-8: Spectrogram color-to-amplitude spectrum.

Also see “Spectrogram controls” on page 59.

Logarithmic or Linear X-Axis Scale

The x-axis defaults to a logarithmic scale, but can be changed to a linear scale if desired. In the View controls (Figure 10-7), click *Logarithmic* to access

the x-axis scale options menu. With a linear scale selected, frequency is constant, but the width of each octave along the x-axis is different. With a logarithmic scale selected, octaves are displayed with a constant width, but frequency is displayed logarithmically within each octave.

Axes display

The *Axes* control (Figure 10-7) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

Pausing the display

The Pause button in the upper right corner of the View section (Figure 10-7) allows you to freeze the display at any time. To resume, click the button again.

Horizontal controls (frequency axis)

The *Horizontal* controls (Figure 10-7) configure the value range of the x-axis (frequency). Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the Horizontal control menu (Figure 10-9).

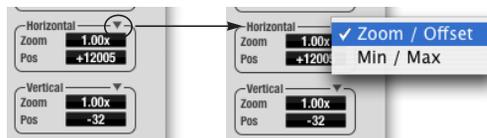


Figure 10-9: Horizontal control menu

In *Zoom/Offset* mode, *Zoom* sets the display zoom from 1x to 100x, where the number represents the zoom factor relative to the entire frequency range. For example, when the horizontal zoom value is 1x, the entire frequency range from 10 to 24000 Hertz is displayed; when the horizontal zoom value is 2x, one half of the entire frequency range is displayed. *Pos* determines which frequency is displayed at the center of the graph.

In *Min/Max* mode, *Min* and *Max* set the lowest and highest displayed frequencies (in Hertz).

Vertical controls (amplitude axis)

The *Vertical* controls (Figure 10-7) operate similarly to the *Horizontal* controls, except that they configure the y-axis (amplitude).

In *Zoom/Offset* mode, *Zoom* sets the display zoom from 1x to 100x, and *Pos* sets the center amplitude of the graph.

In *Min/Max* mode, *Min* and *Max* set the smallest and largest displayed amplitude.

Spectrogram controls

The *Floor* control (Figure 10-5) sets the amplitude threshold for the spectrogram display, from -144 dB up to 1 dB.

The *Alpha* control (Figure 10-5) sets the opacity of the spectrogram information displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

OSCILLOSCOPE

The Oscilloscope (Figure 10-10) graphs the amplitude of an audio signal over time.

Amplitude is displayed on the y-axis and time is displayed on the x-axis. A thick white vertical line marks where time equals zero; a thick white horizontal line marks where amplitude equals zero (Figure 10-10, below).

Level meters are displayed to the right of the graph. One or two meters are shown, depending on the current view mode (see “View controls”).

Opening the oscilloscope

Each MOTU audio interface has its own oscilloscope. To open an oscilloscope, choose the *Oscilloscope* item from the *Devices* menu under the desired interface (Figure 10-6).

Choosing a channel to display

The oscilloscope follows the currently chosen *Scope channels* (Figure 10-1 on page 52).

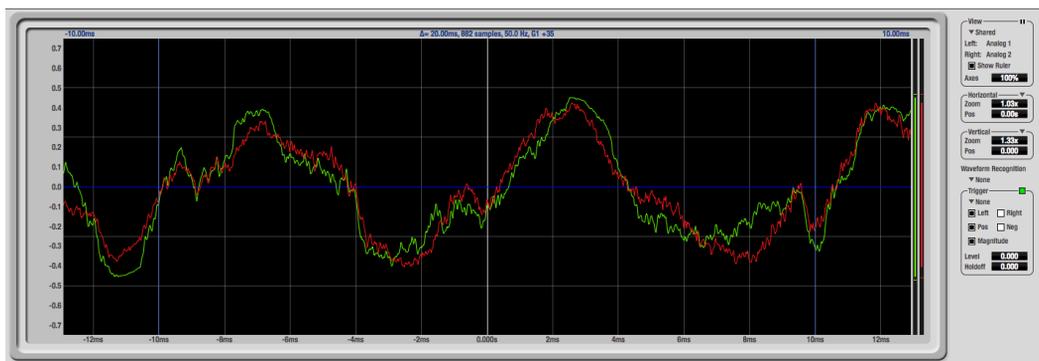


Figure 10-10: Oscilloscope

View controls

The View controls (Figure 10-11) provide several options for the oscilloscope display.

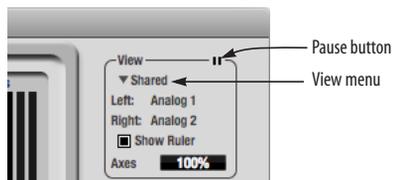


Figure 10-11: View controls

View menu

The View menu (Figure 10-11) lets you choose how you wish to display the audio channel(s) being displayed.

View menu setting	What it displays
Left	Left channel only
Right	Right channel only
Split screen	Left channel on top; right channel on the bottom
Shared	Left and right on top of each other; left is green, right is red
Add	Left and right channels' amplitudes are added together
Subtract L-R	The right channel's amplitude is subtracted from the left channel's amplitude

Display options

The *Axes* control (Figure 10-11) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden). The *Show Ruler* option toggles the measurement items (see “Measurement information” on page 62).

Pausing the display

The Pause button in the upper right corner of the View section (Figure 10-11) allows you to freeze the display at any time. To resume, click the button again. The level meters will remain active while the display is paused.

Horizontal controls (time axis)

The *Horizontal* controls (Figure 10-12) configure the value range of the x-axis (time). Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the Horizontal control menu (Figure 10-12).

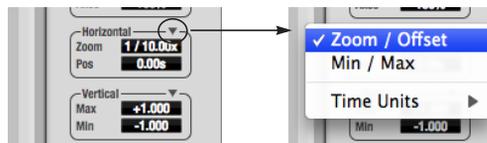


Figure 10-12: Horizontal control menu

In *Zoom/Offset* mode, *Zoom* sets the display zoom from 1/1000x to 10x, where the number represents the number of pixels per sample. For example, when the horizontal zoom value is 10x, 10 samples are displayed in 100 pixels; when the horizontal zoom value is 1/10x, 100 samples are displayed in 10 pixels. *Offset* moves the line marking time equals zero left or right.

In *Min/Max* mode, *Min* and *Max* set the earliest and most recent displayed time.

Time Units

The *Time Units* sub-menu (Figure 10-12) provides the option to view the X axis in Seconds or Samples.

Vertical controls (amplitude axis)

The *Vertical* controls (Figure 10-12) operate similarly to the Horizontal controls, except that they configure the y-axis (amplitude).

In *Zoom/Offset* mode, *Zoom* sets the display zoom from 1/2 to 200x, and *Offset* moves the line marking amplitude equals zero line up or down.

In *Min/Max* mode, *Min* and *Max* set the smallest and largest displayed amplitude.

Waveform Recognition

The Waveform Recognition option searches through new audio data looking for a waveform which most resembles that which was previously displayed. The region where this takes place is a small window around the line marking time equals zero, denoted by the extra vertical graph lines surrounding it. There are two kinds of waveform recognition available: Type I and Type II.

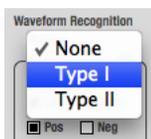


Figure 10-13: Waveform Recognition menu

Type I recognition provides the most stable display of the waveform. It is the most resistant to change. Louder transients, such as those produced by a snare drum, are not displayed inside of the waveform window. Type I is best for observing the shape of a signal produced by a synthesizer or observing the tone of a guitar through a chain of pedals.

Type II recognition is less resistant to change. It will include loud transients within the waveform recognition window. Type II is better for observing percussive music where the beat itself is to be centered within the waveform window.

Trigger

When the *Trigger* (Figure 10-14) is not enabled (the Trigger menu is set to *None*), the graph updates based on time: after every n samples of the monitored audio signal, the most recent samples are displayed. When the Trigger is enabled (set to any mode other than *None*), the graph updates in response to specific conditions in the signal. The Trigger section defines that criteria and how the graph will display the events that match.

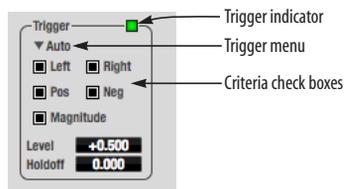


Figure 10-14: Trigger settings

Criteria

The criteria check boxes (Figure 10-14) determine the conditions that the trigger is looking for and where it will look for them.

The *Left* check box causes the condition to be looked for in the left channel of the signal; likewise, the *Right* check box looks for the condition in the right channel. One or both of these can be enabled simultaneously. If neither is enabled, the criteria will not be found because the trigger is not looking at any audio signal.

The *Pos* and *Neg* check boxes determine the slope of the event. When the *Pos* check box is enabled, the trigger will look for an event where amplitude is increasing; likewise, enabling the *Neg* check box tells the trigger to look for an event where amplitude is decreasing. One or both of these can be enabled simultaneously. If neither is enabled, the criteria will not be found because the trigger is not looking for any particular kind of event.

The *Level* setting defines the amplitude threshold that the trigger is looking for. The Level is indicated on the graph by a blue horizontal line (or two blue horizontal lines, if *Magnitude* is enabled). Events which cross this threshold using the enabled slope(s) in the enabled channel(s) will activate the trigger. The response of the trigger is set by the Trigger mode (see “Trigger modes”, below).

Enabling the *Magnitude* check box tells the trigger to look for both positive and negative Level values, regardless of whether the Level value is positive or negative. For example, if Level is set to +0.500 and

Magnitude is enabled, the trigger will look for both +0.500 and -0.500. You will see a second blue line appear in the display when *Magnitude* is enabled to denote the second value.

Holdoff

Holdoff defines a time interval during which the oscilloscope does not trigger. The most recent trace will be displayed during that period. When the period is over, the trigger is “re-armed”, i.e. it will begin looking for the criteria again.

Click and drag this value up or down to set it, or double-click to return to the default value.

Trigger modes

The Trigger menu (Figure 10-14 on page 61) provides four modes:

Trigger mode	What it does
None	The Trigger is not active; this is the default mode. The incoming audio signal will be displayed continuously as audio is received.
Auto	The display is always updating, but when the condition is met, the trigger event will be displayed centered around the line marking time equals zero.
Normal	The display updates only when the condition is met; the last trace will be displayed until the next matching event is found.
Single Sweep	Similar to Normal mode, but the last trace will be displayed until you manually arm the trigger by clicking the Trigger indicator (Figure 10-14 on page 61) or by pressing the spacebar.

Trigger indicator

The Trigger indicator (Figure 10-14 on page 61) displays the state of the trigger, and also provides a way to manually interact with it. The Trigger indicator always displays one of three colors:

Color	Status
Green	When the current Trigger criteria has been met (including when the Trigger mode is <i>None</i>).
Yellow	When the Trigger is armed, but has not yet found an event which matches its criteria. Yellow can also indicate that the graph has been manually paused using the Pause button in the View section (see “Pausing the display” on page 60).
Red	When the Trigger is being held off, either because the Trigger mode is set to Single Sweep or the Holdoff time is not set to zero.

You can also click on the Trigger indicator to force certain actions, depending on the Trigger mode. In Auto and Normal modes, clicking on the Trigger indicator causes the display to run freely; you may click & hold to force this to occur for as long as you’d like. In Single Sweep mode, clicking on the Trigger indicator re-arms the trigger. When the Trigger mode is *None*, clicking on the Trigger indicator has no effect.

Measurement information

You can view detailed information about a particular time range by using the measurement bars.

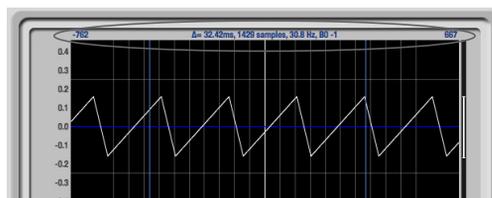


Figure 10-15: Measurement information

To adjust the left and right edges of the measurement area, click and drag the blue bars in the graph, or click and drag the blue numbers in the upper left or right corners. To reset them to the default value, double-click the numbers.

Information about the measured area is displayed at the center of the top ruler: the duration (in seconds and samples), the approximate frequency, and the scientific note name. If the measured area is long enough, the approximate beats per minute (bpm) is displayed.

Ideas for using the Oscilloscope

The Oscilloscope can be used in many useful ways during the routine operation of your recording studio. Here are just a few examples.

Analyzing and comparing harmonic content

The oscilloscope lets you “see” the nature of the harmonic profile in any audio material. You can also view two signals side by side (in stereo mode) to compare their profiles and, if necessary, make adjustments to the source of each signal and view your changes in real time.

Viewing transients such as drum hits

If you loop a snare hit or other similar transient audio clip and feed it through the oscilloscope, you can more or less “freeze” the transient waveform in the oscilloscope frame. This can be useful, for example, for viewing the results of real-time compression that you are applying with an effects plug-in, as demonstrated in Figure 10-16. In this example, a snare hit is being compressed by Digital Performer’s Dynamics plug-in. As you make adjustments to the compression plug-in’s settings, you can see the transient waveform change the next time the Oscilloscope triggers. For compression, this can be particularly useful for balancing the effect of the attack on the transient, relative to the decay portion of the waveform. Conversely, you can see the effect of the threshold setting directly on the decay portion, relative to the attack. In effect, you can see as well as hear the results of your compression adjustments.

To view a transient waveform in the Oscilloscope display, turn off Waveform Recognition and use the *Normal* Trigger mode. Adjust the level high enough to encompass the vertical amplitude of most of the transient. If the transient pulse sweeps across the

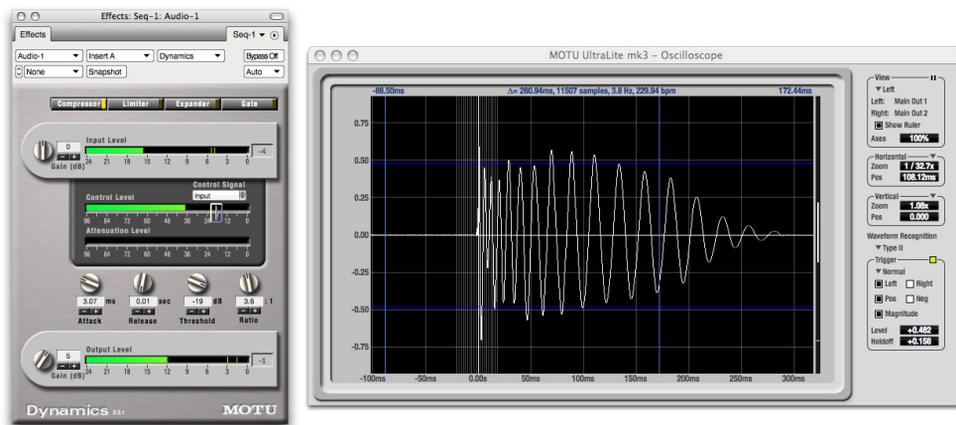


Figure 10-16: Viewing transients in the Oscilloscope

screen, try raising the Holdoff level. Once the transient is settled in the display and fairly stable, you may need to adjust the horizontal position to center it in the display. These settings are depicted in the example in Figure 10-16.

You can also pause the display at any time and adjust the horizontal bounds to locate a transient.

Clip detection

You can use the Oscilloscope to detect clipping in a digital audio signal. To do so, enable all criteria (Figure 10-14 on page 61), choose *Single Sweep* from the trigger menu (Figure 10-14), set the level to 0.999 and click the trigger indicator (Figure 10-14) to arm it (yellow). As soon as the signal clips, the trigger indicator will turn red, and the display will show the offending clip at the line marking time equals zero.

Viewing timing pulses

If you have two audio signals with recognizable, timed pulses in them, and you wish to compare their timing with respect to each other, you can use Split Screen or Shared view to visually compare the timing of the two signals. You can zoom in to the sample level for sample accurate viewing.

Building synthesizer patches

If you are building a synth patch on a synthesizer (or forming similar highly periodic audio material), you can run the audio signal through the Oscilloscope as you adjust its sound to check in real time for undesirable (and possibly inaudible) characteristics, which are easily seen in the Oscilloscope display. A good example is DC offset. If a signal develops DC offset, the apparent vertical center of its overall waveform will drift above or below the line marking amplitude equals zero. Try setting Waveform Recognition to *Type I* and setting Trigger to *None*.

Another example is waveform polarity. If you are combining several raw waveforms, polarity is a critical, yet not always obvious, factor in determining the resulting sound. You can use the Oscilloscope to easily view and compare polarities to see if they are inverted from one another or not. The *Add* and *Subtract L - R* View menu settings are particularly useful here.

You can also use the Oscilloscope to help you apply waveform modulation and keep it “in bounds”. For example, you could easily see if pulse width modulation is collapsing in on itself to choke the sound, an effect that is readily seen in the Oscilloscope display but not necessarily easy to determine by ear when using multiple modulation sources.

Guitarists can also visually observe the effects of their pedals and processing, while playing. With the Trigger mode set to *None* and Waveform Recognition set to *Type I*, the waveform will be tracks automatically.

When applying filters and filter resonance, the visual effect on the waveform can be invaluable in reinforcing what you are hearing as you make adjustments.

Monitoring control voltage output from Volta
MOTU’s Volta instrument plug-in for Mac OS X turns your audio interface into a control voltage interface, giving you precise digital control from your favorite audio workstation software of any hardware device with a control voltage (CV) input. The CV signals output from Volta can be monitored in the Oscilloscope, giving you visual feedback on LFOs, envelopes, ramps, step sequencers, and more.

For more information on Volta, see www.motu.com.

X-Y PLOT

The *X-Y Plot* window (Figure 10-17) graphs the amplitude of a stereo audio signal on a two-dimensional grid.

For each unit of time (i.e. each sample), the amplitude of the left channel is displayed on the x-axis and the amplitude of the right channel is displayed on the y-axis. A thick white vertical line marks where left channel amplitude equals zero; a thick white horizontal line marks where right channel amplitude equals zero (Figure 10-17, below). There are also thick white diagonal lines for $y = x$ and $y = -x$.

Metering

Level meters are displayed above and to the right of the graph for the left (green) and right (red) channels, respectively. An additional *Correlation meter* (blue) is displayed on the right. This meter displays the correlation between the two channels.

The higher the meter, the higher the correlation between the two channels. Below are a few examples:

Situation	Meter level	X-Y Plot graph	Mathematical relationship
Perfect correlation	+1	Diagonal line going from lower left to upper right:	$y = x$
Zero correlation	0	No discernible pattern	None
Perfectly out of phase	-1	Diagonal line going from upper left to lower right:	$y = -x$

Opening the X-Y Plot

Each MOTU audio interface has its own X-Y Plot window. Choose the *X-Y Plot* item from the Devices menu under the desired interface (Figure 10-6 on page 57).

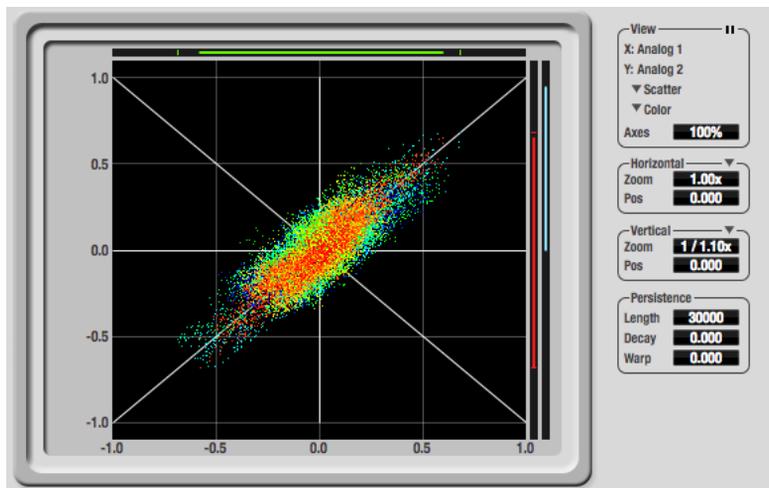


Figure 10-17: X-Y Plot

Choosing a channel to display

The X-Y Plot follows the currently chosen *Scope channels* (Figure 10-1 on page 52).

View controls

The View controls (Figure 10-18) provide several options for the X-Y Plot display.

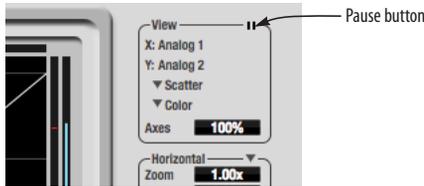


Figure 10-18: View controls

Pausing the display

The Pause button in the upper right corner of the View section (Figure 10-18) allows you to freeze the display at any time. To resume, click the button again. The level meters will remain active while the display is paused.

Line/Scatter

Choose either *Line* or *Scatter* from the menu in the View section (Figure 10-18) to plot each point (sample) as either a single pixel or as a continuous line that connects each plot point to the next, as shown below in Figure 10-19.

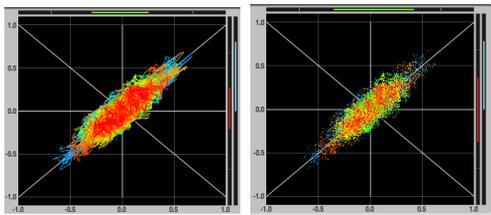


Figure 10-19: The same X-Y Plot displayed in Line versus Scatter mode.

☛ Line mode is significantly more CPU intensive than Scatter. You can reduce Line mode CPU overhead on the X-Y Plot by reducing the Length parameter (described below).

Color/Grayscale

In *Color* mode (Figure 10-18) the most recently displayed audio data is shown in red, which fades to yellow, green and then finally blue, before disappearing. In Grayscale mode, data is first shown in white and then fades to gray. To adjust the scale of this color/brightness change, see “Decay” on page 67.

Axes

The *Axes* control (Figure 10-18) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

Horizontal and vertical controls

The *Horizontal* and *Vertical* controls (Figure 10-20) configure the value range of the x-axis (left channel amplitude), and y-axis (right channel amplitude), respectively. Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the menu shown in Figure 10-20.

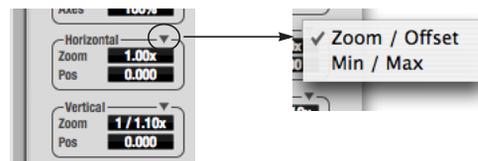


Figure 10-20: Setting the Horizontal or Vertical control modes.

In *Zoom/Offset* mode, *Zoom* scales the axis. *Pos* moves the lines marking $x = 0$ left and right, or $y = 0$ up and down.

In *Min/Max* mode, *Min* and *Max* let you scale the grid by moving the -1.0 and $+1.0$ points along the axis. *Min/Max* mode lets you control the graph boundaries directly.

Persistence

The *Persistence* controls (Figure 10-21) affect the appearance of data from when it is first displayed until it disappears from the grid.

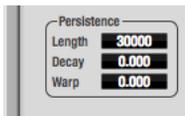


Figure 10-21: The Persistence controls.

Length

Length (Figure 10-21) sets the number of recent samples to show on the plot. For example, when Length is set to 10,000, the 10,000 most recent samples are shown.

Decay

The brightness (in Grayscale mode) or hue (in Color mode) of each sample on the plot is determined by a linear scale, with the most recent sample displayed at the maximum value and the oldest sample displayed at the minimum value.

Decay (Figure 10-21 on page 67) determines the brightness or hue of the minimum value. When Decay is zero, the oldest sample is black. When Decay is +1.000, the oldest sample is fully opaque (in Grayscale mode) or red (in Color mode).

Warp

Warp (Figure 10-21) determines the position of data points after they are first drawn. When warp is zero, data points remain in the same position.

When warp is positive, they contract towards the origin (center of the grid). When warp is negative, they expand away from the origin. The further the warp value is from zero, the greater the effect.

Using the X-Y Plot

The X-Y Plot helps you “see” the width of the stereo field of a mix. It also helps you determine if a mix has issues with polarity, as follows:

Activity on the X-Y Plot	What it indicates
Signal activity occurs mostly along the $x = y$ axis (lower left to upper right) and the Correlation meter reading is high	Left and right channels are predominantly in polarity (the stereo field is relatively narrow)
Signal activity occurs mostly along the $y = -x$ axis (upper left to lower right) and the Correlation meter reading is low (near -1)	Left and right channels are predominantly out of polarity (not in phase)
Signal activity occurs in a seemingly random fashion throughout the grid	No phase relationship exists (i.e. it is probably a wide stereo field)

If a stereo signal is out of phase, it is not mono compatible because it can cancel itself out, either partially or nearly completely, when collapsed to mono.

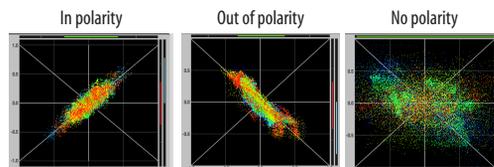


Figure 10-22: Checking polarity in a stereo signal with the X-Y Plot.

PHASE ANALYSIS

The *Phase Analysis* window (Figure 10-23 on page 68) graphs frequency versus phase difference versus amplitude of a stereo signal on either rectangular or polar coordinates.

In rectangular coordinates, the vertical axis represents frequency, and the horizontal axis represents the phase of the left channel minus the phase of the right channel (measured in radians).

In polar coordinates, the radius represents frequency and the angle (theta) from the +y vertical axis represents the phase difference of left channel minus the right channel.

Correlation Meter

The blue *Correlation Meter* to the right of the display shows the correlation between the two channels. The higher the meter, the higher the correlation between the two channels.

Opening the Phase Analysis

Each MOTU audio interface has its own Phase Analysis window. Choose the *Phase Analysis* item from the Devices menu under the desired interface (Figure 10-6 on page 57).

Choosing a channel to display

The Phase Analysis window follows the currently chosen *Scope channels* (Figure 10-1 on page 52).

View controls

The View controls (Figure 10-24) provide several options for the Phase Analysis display.

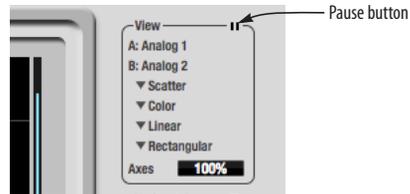


Figure 10-24: View controls

Pausing the display

The Pause button in the upper right corner of the View section (Figure 10-24) allows you to freeze the display at any time. To resume, click the button again. The correlation meter will remain active while the display is paused.

A/B (stereo audio channels)

The *View* section (Figure 10-24) displays the pair of input or output audio channels you are viewing. See “Choosing a channel to display” above.

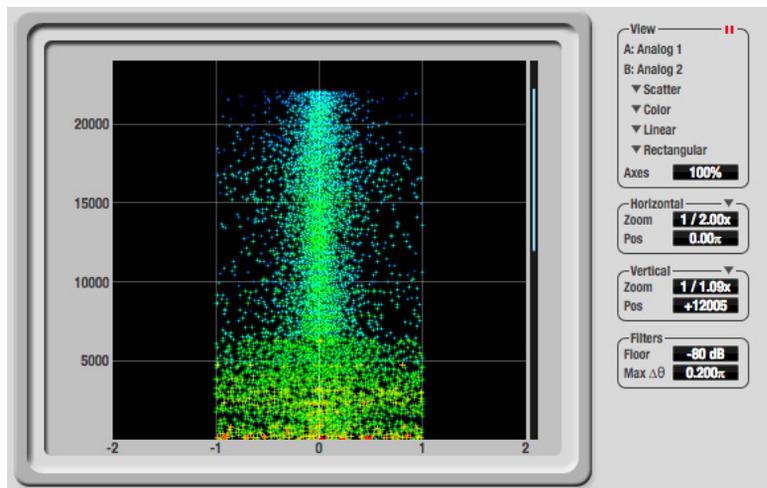


Figure 10-23: Phase Analysis

Line/Scatter

Choose either *Line* or *Scatter* from the menu in the View section (Figure 10-24) to plot each data point as either a single pixel or as a continuous line that connects each frequency data point to the next, as shown below in Figure 10-19.

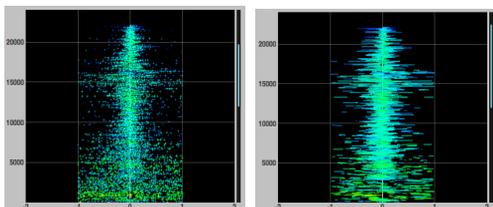


Figure 10-25: The same Phase Analysis displayed in Line versus Scatter mode.

☛ Line mode is significantly more CPU intensive than Scatter. You can reduce Line mode CPU overhead for the Phase Analysis display by increasing the Floor filter and reducing the Max Delta Theta filters (see “Filters” on page 70).

Color/Grayscale

In *Color* mode (Figure 10-24) signal amplitude is indicated by color as follows: red is loud and blue is soft. In grayscale mode, white is loud and gray is soft.

Linear/Logarithmic

Choose either *Linear* or *Logarithmic* from the menu in the View section (Figure 10-24) to change the scale of the frequency axis. In rectangular coordinates, the vertical axis represents frequency, and in polar coordinates, the radius from the center is frequency. With a linear scale, frequencies are spaced evenly; in a logarithmic scale, each octave is spaced evenly (frequencies are scaled logarithmically within each octave).

Linear is better for viewing high frequencies; logarithmic is better for viewing low frequencies.

Rectangular/Polar

Choose either *Rectangular* or *Polar* from the menu in the View section (Figure 10-24) to control how audio is plotted on the Phase Analysis grid.

Rectangular plots the audio on an X-Y grid, with frequency along the vertical axis and phase difference on the horizontal axis. *Polar* plots the data on a polar grid with zero Hertz at its center. The length of the radius (distance from the center) represents frequency, and the angle (theta) measured from the +y (vertical) axis represents the phase difference in degrees.

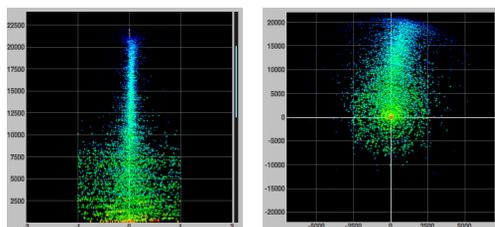


Figure 10-26: Rectangular versus Polar display (with a linear plot).

Above, Figure 10-26 shows Rectangular versus Polar display with a Linear plot. Below, Figure 10-27 shows the same displays (and the same data) with a Logarithmic plot:

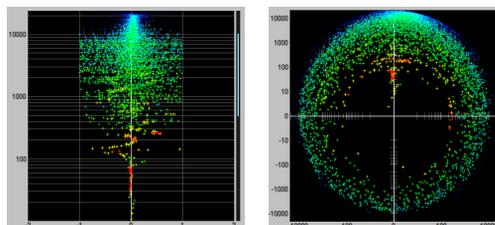


Figure 10-27: Rectangular versus Polar display with a logarithmic plot.

Axes

The *Axes* control (Figure 10-24) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

Horizontal and vertical controls

The *Horizontal* and *Vertical* controls (Figure 10-28) let you scale each axis of the grid and offset its zero point. Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the menu shown in Figure 10-28.

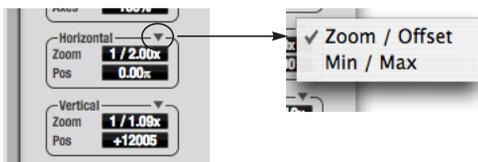


Figure 10-28: Setting the Horizontal or Vertical control modes.

In *Zoom/Offset* mode, *Zoom* scales the axis. *Pos* moves the zero line.

In *Min/Max* mode, *Min* and *Max* let you scale the grid by moving the end points along the axis. *Min/Max* mode lets you set the boundaries of the graph directly.

Filters

The *Filters* section (Figure 10-29) lets you control the density of the Phase Analysis display.

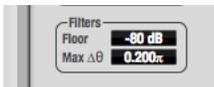


Figure 10-29: Filters

Floor

Floor (Figure 10-29) determines the amplitude threshold for the display. When the amplitude of both channels drops below this threshold, the signal is not shown.

Max delta theta

Max delta theta (Figure 10-29) only affects Line view (see “Line/Scatter” on page 69) and sets the maximum difference in frequency between plot

points in the line plot. For two adjacent frequencies, if the distance (phase difference) between the two frequencies is greater than the Max delta theta, then the line is not drawn.

Using the Phase Analysis

In the polar display (top row of Figure 10-30 on page 71), stereo material that is predominantly phase-aligned (correlated) appears along the vertical axis, as demonstrated in the first column (*Perfectly in phase*) in Figure 10-30. If the vertical line tilts left or right, this indicates general differences in phase; the more the tilt (delta theta), the more the phase difference. If the vertical line points downwards in the polar display, this indicates that the stereo image is predominantly out of polarity, as demonstrated by the fourth column (*Inverted*) in Figure 10-30. Delays appear as spirals in the polar display.

The rectangular display (bottom row of Figure 10-30) also shows a predominantly phase-aligned stereo image along the vertical axis, and tilt (or left-right offset) from the center vertical axis represents differences in phase. If a signal is predominantly out of polarity, it appears along the $\theta = -1.0$ or $\theta = +1.0$ lines in the rectangular display, as demonstrated in the fourth column (*Inverted*) in Figure 10-30 on page 71.

Using Phase Analysis for multiple mic placement

The polar display can be very useful when recording drums or another instrument with multiple microphones. The slight delays caused by the differences in distance to the source can often create a comb filtering (delay) effect between two mic signals, due to phase cancellation. These comb filter effects appear as spirals in the polar display. If you arrange the mics so that the null points (where the spiral pattern meets the negative y axis) are

outside the critical frequency range of the instrument being recorded, you can avoid phase problems among the mic signals.

Tuning PA systems

The Phase Analysis window can also be used to troubleshoot and tune PAs and sound reinforcement systems by placing microphones in strategic locations, comparing the two signals in the Phase Analysis grid and looking for phase issues at various locations.

Summing to mono

The Phase Analysis window is ideal for checking stereo audio that needs to be summed to mono. The Phase Analysis lets you see what frequencies will be canceled out when summed.

In the rectangular view, any lines in the signal that touch the +1.0 or -1.0 vertical lines in the grid will be canceled out at the frequency where they touch, when the signal is summed to mono.

In the polar view, any signal that falls on the negative y axis (below zero) in polar view will be canceled out when the signal is summed to mono.

Checking for phase issues in stereo tracks

You can use the Phase Analysis window to check the overall polarity of a stereo mix. Figure 10-31 is an example of a full stereo mix that has phase issues, as indicated by the majority of the signal's energy, which is predominantly skewed to the left side of the rectangular view (left) and spread along the -y axis in the polar view (right).

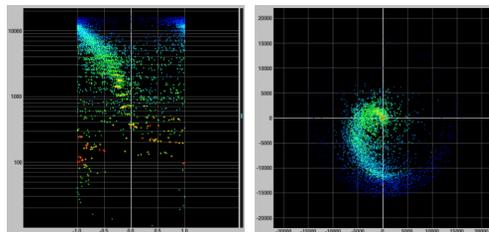


Figure 10-31: A stereo mix with phase issues.

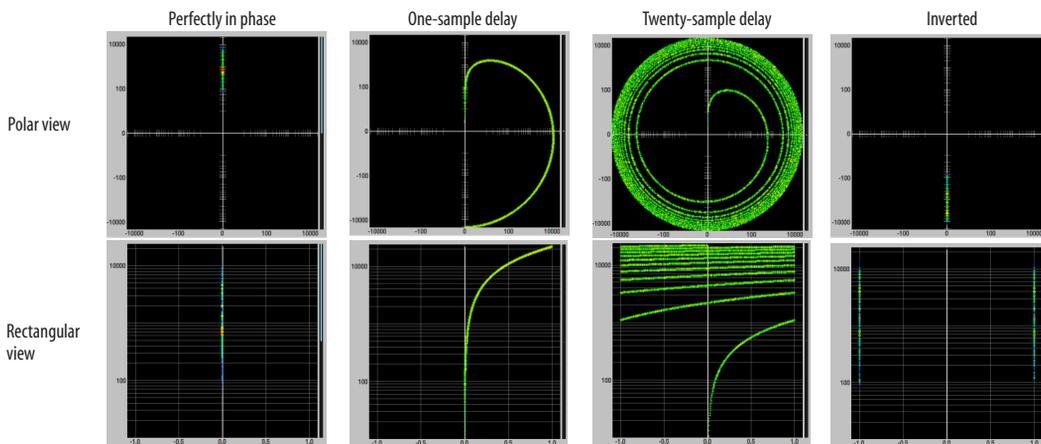
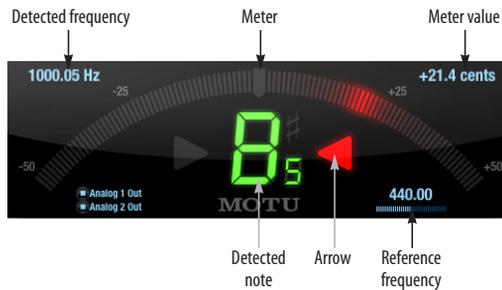


Figure 10-30: Two identical audio streams in the Phase Analysis.

TUNER



The *Tuner* window is an accurate and easy to use tuner.

Opening the Tuner

Each MOTU audio interface has its own Tuner window. Choose the *Tuner* item from the Devices menu under the desired interface (Figure 10-6 on page 57).

Choosing a channel to tune

The Tuner follows the currently chosen *Scope channels* (Figure 10-1 on page 52).

In the Tuner window, the displayed channel pair is shown in the lower left corner. Each channel has a checkbox to enable or disable its input to the tuner.

Tuner controls

Detected frequency: fundamental frequency of the incoming signal, in Hertz (Hz).

Detected note: note name and octave that correspond to the detected fundamental frequency.

Meter: representation of the pitch difference between the detected note and the detected fundamental frequency. The horizontal position of the illuminated segments indicates how far the detected frequency is from the detected note. The number of illuminated segments indicates uncertainty or inharmonicity in the signal; a greater number of illuminated segments represents

greater uncertainty. The color of the segments changes gradually from green (in tune) to yellow, orange, and red (progressively further out of tune).

Meter value: difference between the detected note and the detected frequency, in cents.

Arrows: the direction in which the detected frequency needs to move to match the frequency of the detected note. The color of the arrows changes progressively in the same manner as the meter segments. When the detected fundamental frequency matches the detected note within three cents, both arrows will be illuminated.

Reference frequency: sets the frequency reference for the pitch A4, between 400 and 480 Hz. The default frequency is 440 Hz. The reference frequency can be adjusted by dragging on the bar below the number, or by clicking the number and typing a value. To reset the tuner to the default frequency, double-click the slider, or click the number, press the Delete key, and press Enter.

Tuning stereo signals

When tuning a stereo signal, the tuner analyzes the sum of the two channels. If the channels are not phase coherent, the tuner may not be able to measure the frequency of the signal. To tune only one channel of the channel pair, disable one of the channels as described in “Choosing a channel to tune”.

CONFIGURATIONS MENU

A configuration is just like a hardware preset (a “snapshot” of all settings in CueMix FX and therefore the Audio Express hardware itself), except that it can be created and managed using the CueMix FX software on your computer, completely independently of the Audio Express hardware. The commands in the Configurations menu let you create, save, load, import, export and otherwise manage as many configurations as you wish.

Here is a summary of Configurations menu operations:

Configurations Menu item	What it does
Create New	Lets you name and save a new configuration, which appears at the bottom of the Configurations menu.
Save	Overwrites the current configuration (checked in the list at the bottom of the menu) with the current settings in CueMix FX.
Save To	Same as Save above, except that it lets you first choose the configuration you wish to save to (instead of the current one).
Delete	Lets you choose a configuration to permanently remove from the menu.
Import	Loads all configurations from a configuration file on disk.
Export	Saves all current configurations as a file on disk.
Configuration list	Choose any configuration to load it. The current (last loaded or saved) configuration has a check mark next to it.

Modifying a configuration

The name of the current configuration is displayed in the CueMix FX window title bar. If you make any changes to the settings in CueMix FX, an asterisk appears in front of the name to remind you that the current state of CueMix FX doesn't match the saved configuration. If you wish to update the saved configuration with the new changes, use the Save command. To save the current state of CueMix FX to another configuration, use Save To. To save as a new, separate configuration, use Create New.

Saving a CueMix FX configuration as a hardware preset

To save a CueMix FX configuration as a hardware preset:

- 1 Choose the configuration from the Configurations menu to make it the current active configuration.
- 2 Choose *File menu* > *Save Hardware Preset*.

- 3 Type in a name, choose a preset slot and click OK.

Saving a hardware preset as a CueMix FX configuration

To save a hardware preset as a CueMix FX configuration:

- 1 Choose *File menu* > *Load Hardware Preset* to make it the current active preset.
- 2 Choose *Configuration menu* > *Create New* (or *Save To*) to save it as a configuration.

TALKBACK MENU

Choose the commands in the Talkback menu to engage or disengage Talkback or Listenback. See "Talkback and listenback" on page 54.

PHONES MENU

The Phones menu allows you to choose what you will hear on the headphone output, just like the Phones setting in MOTU Audio Console. However, this menu provides one extra option that is exclusive to CueMix FX: *Follow Active Mix*. This menu item, when checked, causes the headphone output to mirror the output of the current mix being viewed in CueMix FX. For example, if you are currently viewing mix bus 3, the headphones will mirror the mix bus 3 output (whatever it is assigned to).

CONTROL SURFACES MENU

CueMix FX can be controlled from an automated control surface such as the Mackie Control™. Use the commands in the *Control Surfaces* menu to enable and configure this feature.

Application follows control surface

When checked, the *Application follows control surface* menu command makes the CueMix FX window scroll to the channel you are currently adjusting with the control surface, if the channel is not visible when you begin adjusting it. The same is true for the bus tabs: if you adjust a control in a bus

that is not currently being displayed, CueMix FX will jump to the appropriate tab to display the control you are adjusting.

Share surfaces with other applications

When the *Share surfaces with other applications* menu command is checked, CueMix FX releases the control surface when you switch to another application. This allows you to control your other software with the control surface. Here's a simple way to understand this mode: the control surface will always control the front-most application. Just bring the desired application to the front (make it the active application), and your control surface will control it. When you'd like to make changes to CueMix FX from the control surface, just bring CueMix FX to the front (make it the active application).

When this menu item is unchecked, your control surface will affect CueMix FX all the time, even when CueMix FX is not the front-most application. In addition, you will not be able to control other host audio software with the control surface at any time (because CueMix FX retains control over it at all times). This mode is useful when you do not need to use the control surface with any other software.

CueMix Control Surfaces

CueMix FX includes support for the following control surface products:

- Mackie Control™
- Mackie HUI™
- Mackie Baby HUI™

Use the sub-menu commands in the *CueMix Control Surfaces* menu item to turn on and configure control surface support, as described briefly below.

Enabled

Check this menu item to turn on control surface operation of CueMix FX. Uncheck it to turn off control surface support.

Configure...

Choose this menu item to configure your control surface product. Launch the on-line help for specific, detailed instructions on configuring CueMix FX for operation with your control surface product.

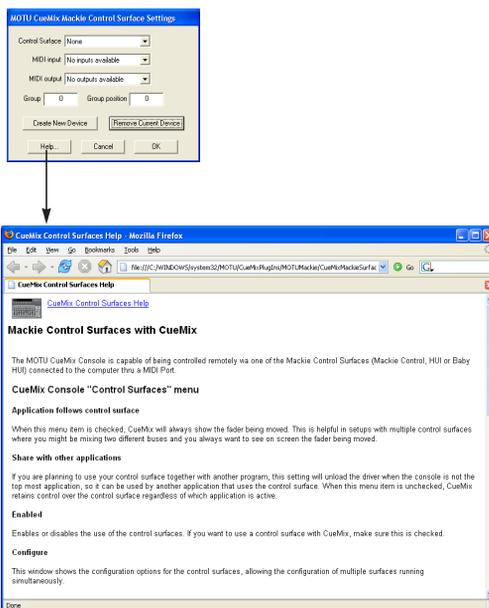


Figure 10-32: Refer to the extensive on-line help for details about configuring CueMix FX for operation with your control surface product.

CHAPTER 11 MOTU SMPTE Console

OVERVIEW

The Audio Express can resolve directly to SMPTE time code via any analog input, without a separate synchronizer. The Audio Express can also generate time code via any analog output. The Audio Express provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy. Direct time code synchronization is supported by Cubase, Nuendo and other audio sequencer software that supports the ASIO2 sample-accurate sync protocol.

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MOTU SMPTE SETUP

The included MOTU SMPTE Setup™ software provides a complete set of tools to generate SMPTE for striping, regenerating or slaving other devices to the computer.

CLOCK/ADDRESS

The *Clock/Address* menu (Figure 11-1) provides the same global *Clock Source* setting as in MOTU Audio Console (“Clock Source” on page 28), but it includes additional information: each setting shows both the clock and the address (time code or sample location), separated by a forward slash (/). To resolve the Audio Express to SMPTE time code, choose the *SMPTE / SMPTE* setting in the *Clock/Address* menu. This means that the system will use SMPTE as the clock (time base) and SMPTE as the address.

FRAME RATE

This setting should be made to match the SMPTE time code frame rate of the time code that the system will be receiving. The Audio Express can

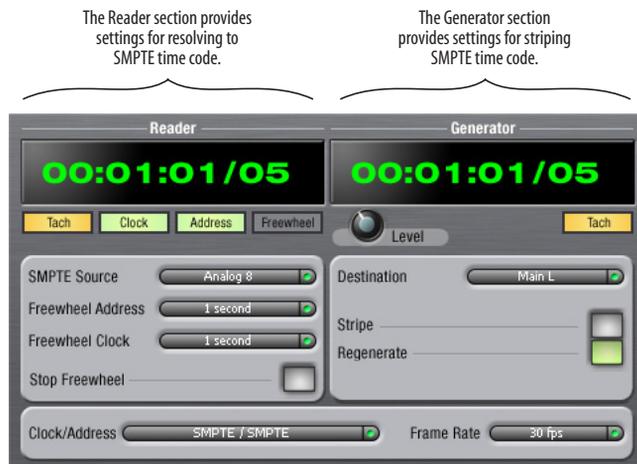


Figure 11-1: SMPTE Setup gives you access to your Audio Express's on-board SMPTE time code synchronization features.

auto-detect and switch to the incoming frame rate, except that it cannot distinguish between 30 fps and 29.97 fps time code, or 23.976 and 24 fps time code. So if you are working with either of these rates, make sure you choose the correct rate from this menu.

READER SECTION

The Reader section (on the left-hand side of the window in Figure 11-1) provides settings for synchronizing the Audio Express to SMPTE time code.

Status lights

The four status lights (Tach, Clock, Address and Freewheel) give you feedback as follows.

Tach

The Tach light blinks once per second when the Audio Express has successfully achieved lockup to SMPTE time code and SMPTE frame locations are being read.

Clock

The Clock light glows continuously when the Audio Express has successfully achieved lockup to an external time base, such as SMPTE time code or the S/PDIF input.

Address

The Address light glows continuously when the Audio Express has successfully achieved lockup to SMPTE time code.

Freewheel

The Freewheel light illuminates when the Audio Express is freewheeling address (time code), clock or both. For details about Freewheeling, see “Freewheel Address” and “Freewheel clock” below.

SMPTE source

Choose the analog input that is connected to the time code source. This is the input that the Audio Express “listens” to for time code.

Freewheel Address

Freewheeling occurs when there is a glitch or drop-out in the incoming time code for some reason. The Audio Express can freewheel past the drop-out and then resume lockup again as soon as it receives readable time code. Choose the amount of time you would like the Audio Express to freewheel before it gives up and stops altogether.

The Audio Express cannot freewheel address without clock. Therefore, the *Freewheel Address* setting will always be lower than or equal to the *Freewheel Clock* setting, and both menus will update as needed, depending on what you choose.

Keep in mind that freewheeling causes the system to keep going for as long as the duration you choose from this menu, even when you stop time code intentionally. Therefore, if you are starting and stopping time code frequently (such as from the transports of a video deck), shorter freewheel times are better. On the other hand, if you are doing a one-pass transfer from tape that has bad time code, longer freewheel times will help you get past the problems in the time code.

The ‘Infinite’ freewheel setting

The *Infinite* freewheel setting in the *Freewheel Address* menu causes the Audio Express to freewheel indefinitely, until it receives readable time code again. To make it stop, click the *Stop Freewheeling* button.

Freewheel clock

Freewheeling occurs when there is glitch or drop-out in the incoming SMPTE time code for some reason. The Audio Express can freewheel past the drop-out and then resume lockup again as soon as it receives a stable, readable clock signal.

The Audio Express cannot freewheel address without clock. Therefore, the *Freewheel Address* setting will always be lower than or equal to the *Freewheel Clock* setting, and both menus will update as needed, depending on what you choose.

The ‘Infinite’ freewheel setting

The *Infinite* freewheel setting in the *Freewheel Clock* menu causes the Audio Express to freewheel indefinitely, until it receives readable time code again. To make it stop, click the *Stop Freewheeling* button.

Stop Freewheeling

The Stop Freewheeling button stops the system if it is currently freewheeling.

GENERATOR SECTION

The Generator section (on the right-hand side of the window in Figure 11-1) provides settings for generating SMPTE time code.

Level

Turn the level knob to adjust the volume of the SMPTE time code being generated by the Audio Express. The level knob disappears when the Destination is set to *None*.

Tach light

The Tach light blinks once per second when the Audio Express is generating SMPTE time code.

Destination

In the *Destination* menu, choose either *SMPTE* (to generate time code) or *None* (to turn it off).

Stripe

Click this button to start or stop time code. To set the start time, click directly on the SMPTE time code display in the Generator section and type in the desired start time. Or drag vertically on the numbers.

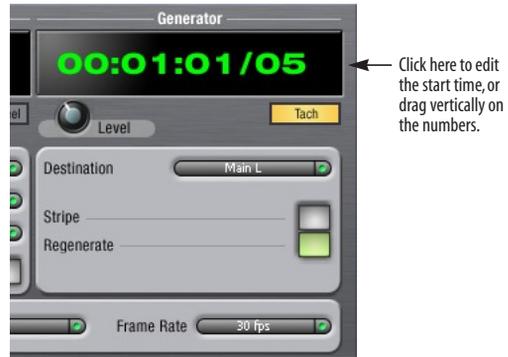


Figure 11-2: Setting the time code start time.

Regenerate

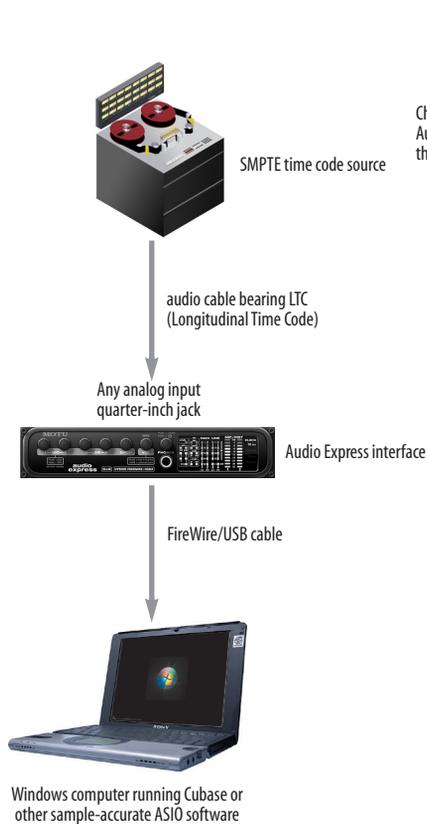
This option, when enabled, causes the generator to generate time code whenever the Audio Express is receiving SMPTE time code.

RESOLVING CUBASE OR NUENDO TO SMPTE TIME CODE

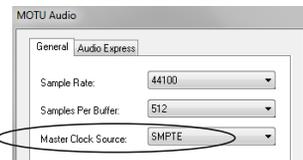
To resolve your Audio Express and Cubase or Nuendo directly to SMPTE time code with no additional synchronization devices, use the setup shown in Figure 11-3 on page 78. Make sure the *Clock Source* setting in the MOTU Audio Console window is set to *SMPTE*. Also, make sure that you’ve connected an LTC input signal to an Audio Express analog input, and that you’ve specified that input in the SMPTE Setup.

SYNCING TO SMPTE TIME CODE

The Audio Express system can resolve directly to SMPTE time code. It can also generate time code, under its own clock or while slaving to time code. Therefore, the Audio Express can act both as an audio interface and as a SMPTE time code synchronizer to which you can slave other devices. You can use the Audio Express to slave your audio software to time code as well, as long as your software supports sample-accurate sync, which is the means by which the software follows the Audio Express. The accuracy may not be sample-accurate, but in most cases it will be very close.



Choose *SMPTE* as the clock source in MOTU Audio Setup. This setting can also be made in the MOTU SMPTE Console (shown below).

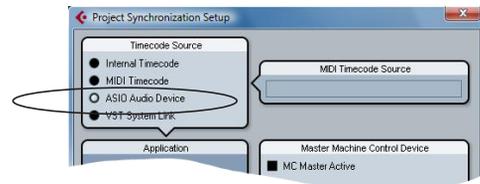


Use this setup if you have:

- ✓ A SMPTE time code source, such as a multitrack tape deck.
- ✓ An Audio Express by itself, OR with another slaved device (such as a digital mixer).
- ✓ Host software that supports sample-accurate sync.

This setup provides:

- ✓ Continuous sync to SMPTE time code.
- ✓ Sub-frame timing accuracy.
- ✓ Transport control from the SMPTE time code source.



In Cubase, go to the Synchronization window and choose *ASIO Audio Device* as the Timecode Base.



Launch the MOTU SMPTE Console and specify the *SMPTE Source*, which is the interface receiving the SMPTE time code. Also, confirm that the *Clock Source/Address* is *SMPTE/SMPTE*.

Figure 11-3: Connections for synchronizing the Audio Express directly to SMPTE time code.

CHAPTER 12 Performance Tips & Troubleshooting

Things seems like they are working fine, but the Audio Express then just drops off line, and the computer can't see it anymore.

This is a common symptom when the problem is that the Audio Express is not getting enough power via its Firewire connection to the computer. Check the power source for the Audio Express. Make sure that the power connection meets the requirements outlined in “Power options” on page 22.

Why does the start sound not play through the Audio Express?

The Audio Express will only playback audio at a sampling rate of 44.1, 48, 88.2, or 96 kHz.

The computer freezes when it starts up

If the computer is unable to boot up, it may be a conflict with the WDM Driver. As soon as the computer boots up, Windows will try and initialize the WDM Driver. If this fails, your computer will hang. To determine if the WDM Driver is the problem, boot up in Safe Mode or remove the MOTU Audio software with the Add/Remove Programs Control Panel then restart. Reinstall the Audio Express software and choose only to install the MOTU ASIO driver, not the WDM Driver. Restart again.

'New hardware detected' window

If you connect your Audio Express before running the Audio Express Software Installer CD, Windows will prompt you that new hardware has been detected. Cancel this, and run the Audio Express Software Installer, rather than allowing Windows to locate the drivers.

Audio Express inputs and outputs are not available in SONAR

Make sure that the inputs and/or outputs that you want to use are enabled in SONAR.

Sound Forge and 'Smooth Scroll'

If the Smooth Scroll option under the Options menu is turned on, you may experience clicks and pops or other anomalies in the audio playback. This gets worse as you zoom in more to the waveform while playing. For the best quality playback, turn this option off.

Clicks and pops due to hard drive problems

If you have checked your clock settings sync cable connections and you are still getting clicks and pops in your audio, you may have a drive related problem. Set the Audio Express Clock Source to Internal and try recording just using the analog inputs and outputs of the Audio Express. If you encounter the same artifacts you may want try using another drive in your computer. Clicks and pops can also occur when the drive is severely fragmented or the disk drivers are outdated.

If you are using a FireWire drive on the same bus as the Audio Express, it could be that the FireWire bus is overloaded (too many devices on the same bus). Try removing all devices except the Audio Express.

Connecting or powering gear during operation

It is not recommended that you connect/disconnect, or power on/off devices connected to the Audio Express while recording or playing back audio. Doing so may cause a brief glitch in the audio.

Monitoring - how do I monitor inputs?

Please refer to the documentation for the audio application that you are using. If your application does not support input monitoring, you will need to use the Audio Express's hardware-based CueMix FX monitoring feature. Please see chapter 9, “Reducing Monitoring Latency” (page 45).

Controlling monitoring latency

See chapter 9, “Reducing Monitoring Latency” (page 45).

CUSTOMER SUPPORT

We are happy to provide customer support to our registered users. If you haven’t already done so, please take a moment to register on line at motu.com, or fill out and mail the registration card included with your 8pre. Doing so entitles you to technical support and notices about new products and software updates.

REPLACING DISKS

If your Audio Express installer CD becomes damaged and fails to provide you with fresh, working copies of the program, our Customer Support Department will be glad to replace it. You can request a replacement disc by calling our business office at (617) 576-2760 and asking for the customer service department. In the meantime, you can download the latest drivers from www.motu.com.

TECHNICAL SUPPORT

If you are unable, with your dealer’s help, to solve problems you encounter with the Audio Express system, you may contact our technical support department in one of the following ways:

- Tech support hotline: (617) 576-3066 (Monday through Friday, 9 am to 6 pm EST)

- Tech support 24-hour fax line: (617) 354-3068
- Online support: www.motu.com/support

Please provide the following information to help us solve your problem as quickly as possible:

- The serial number of the Audio Express system. This is printed on a sticker placed on the bottom of the Audio Express itself. You must be able to supply this number to receive technical support.
- A brief explanation of the problem, including the exact sequence of actions which cause it, and the contents of any error messages which appear on the screen.
- The pages in the manual which refer to the parts of the Audio Express with which you are having trouble.
- The version of your computer’s operating system.

We’re not able to solve every problem immediately, but a quick call to us may yield a suggestion for a problem which you might otherwise spend hours trying to track down.

If you have features or ideas you would like to see implemented, we’d like to hear from you. Please write to the Audio Express Development Team, MOTU Inc., 1280 Massachusetts Avenue, Cambridge, MA 02138.

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