MOTU PCI-424™

User's Guide for Macintosh

Mark of the Unicorn, Inc.
1280 Massachusetts Avenue
Cambridge, MA 02138
Business voice: (617) 576-2760
Business fax: (617) 576-3609
Tech support phone: (617) 576-3066
Tech support fax: (617) 354-3068
Tech support email: techsupport@motu.com
Web site: http://www.motu.com

SAFETY PRECAUTIONS AND ELECTRICAL REQUIREMENTS

WARNING: TO REDUCE THE RISK OF FIRE OR ELECTRICAL SHOCK, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR OTHER MOISTURE.

CAUTION: TO REDUCE THE RISK OF ELECTRICAL SHOCK, DO NOT REMOVE COVER. NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

WARNING: DO NOT PERMIT FINGERS TO TOUCH THE TERMINALS OF PLUGS WHEN INSTALLING OR REMOVING THE PLUG TO OR FROM THE OUTLET.

WARNING: IF NOT PROPERLY GROUNDED YOUR MOTU AUDIO INTERFACE COULD CAUSE AN ELECTRICAL SHOCK.

Your MOTU audio interface is equipped with a three-conductor cord and grounding type plug which has a grounding prong, approved by Underwriters' Laboratories and the Canadian Standards Association. This plug requires a mating three-conductor grounded type outlet as shown in Figure A below.

If the outlet you are planning to use for your MOTU audio interface is of the two prong type, DO NOT REMOVE OR ALTER THE GROUNDING PRONG IN ANY MANNER. Use an adapter as shown below and always connect the grounding lug to a known ground. It is recommended that you have a qualified electrician replace the TWO prong outlet with a properly grounded THREE prong outlet. An adapter as illustrated below in Figure B is available for connecting plugs to two-prong receptacles.

Figure A

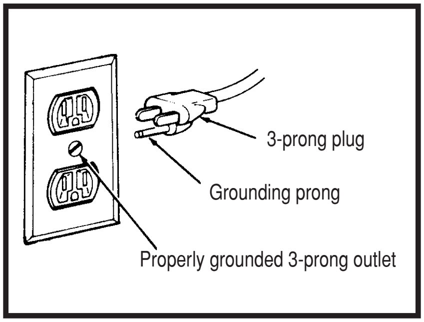
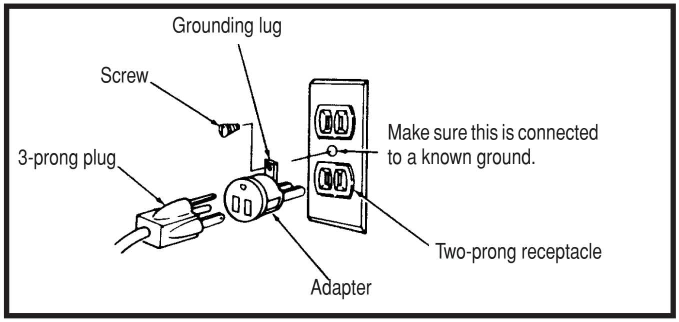


Figure B



WARNING: THE GREEN GROUNDING LUG EXTENDING FROM THE ADAPTER MUST BE CONNECTED TO A PERMANENT GROUND SUCH AS TO A PROPERLY GROUNDED OUTLET BOX. NOT ALL OUTLET BOXES ARE PROPERLY GROUNDED.

If you are not sure that your outlet box is properly grounded, have it checked by a qualified electrician. NOTE: The adapter illustrated is for use only if you already have a properly grounded two-prong receptacle. Adapter is not allowed in Canada by the Canadian Electrical Code. Use only three wire extension cords which have three-prong grounding type plugs and three-prong receptacles which will accept your MOTU audio interface's plug.

IMPORTANT SAFEGUARDS

1. Read instructions - All the safety and operating instructions should be read before operating your MOTU audio interface.
2. Retain instructions - The safety instructions and owner's manual should be retained for future reference.
3. Heed Warnings - All warnings on your MOTU audio interface and in the owner's manual should be adhered to.
4. Follow Instructions - All operating and use instructions should be followed.
5. Cleaning - Unplug your MOTU audio interface from the computer before cleaning and use a damp cloth. Do not use liquid or aerosol cleaners.
6. Overloading - Do not overload wall outlets and extension cords as this can result in a risk of fire or electrical shock.
7. Power Sources - This MOTU interface should be operated only from the type of power source indicated on the marking label. If you are not sure of the type of power supply to your location, consult your local power company.
8. Power-Cord Protection - Power-supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them. Pay particular attention to cords and plugs, convenience receptacles, and the point where they exit from your MOTU audio interface.
9. Lightning - For added protection for your MOTU audio interface during a lightning storm, unplug it from the wall outlet. This will prevent damage to your MOTU audio interface due to lightning and power line surges.
10. Servicing - Do not attempt to service this MOTU interface yourself as opening or removing covers will expose you to dangerous voltage and other hazards. Refer all servicing to qualified service personnel.
11. Damage Requiring Service - Unplug your MOTU audio interface from the computer and refer servicing to qualified service personnel under the following conditions.
    a. When the power supply cord or plug is damaged.
    b. If liquid has been spilled or objects have fallen into your MOTU audio interface.
    c. If your MOTU audio interface has been exposed to rain or water.
    d. If your MOTU audio interface does not operate normally by following the operating instructions in the owner's manual.
    e. If your MOTU audio interface has been dropped or the cabinet has been damaged.
    f. When your MOTU audio interface exhibits a distinct change in performance, this indicates a need for service.
12. Replacement Parts - When replacement parts are required, be sure the service technician has used replacement parts specified by the manufacturer or have the same characteristics as the original part. Unauthorized substitutions may result in fire, electric shock or other hazards.
13. Safety Check - Upon completion of any service or repairs to this MOTU interface, ask the service technician to perform safety checks to determine that the product is in safe operating conditions.

ENVIRONMENT

Operating Temperature: 10°C to 40°C (50°F to 104°)

AVOID THE HAZARDS OF ELECTRICAL SHOCK AND FIRE

Do not handle the power cord with wet hands. Do not pull on the power cord when disconnecting it from an AC wall outlet. Grasp it by the plug.

INPUT

Line Voltage: 100 - 120 volts AC, RMS (US and Japan) or 220 - 250 volts AC, RMS (Europe). Frequency: 47 - 63 Hz single phase. Power: 7 watts maximum.

CAUTION: DANGER OF EXPLOSION IF BATTERY IS REPLACED. REPLACE ONLY WITH THE SAME OR EQUIVALENT TYPE RECOMMENDED BYMANUFAC-TURER. DISPOSE OF USED BATTERY ACCORDING TO MANUFACTURER'S INSTRUCTIONS.

Mark of the Unicorn License Agreement and Limited Warranty on Software

TO PERSONS WHO PURCHASE OR USE THIS PRODUCT: carefully read all the terms and conditions of this agreement before using this software package. Using this software package indicates your acceptance of the terms and conditions of this license agreement.

Mark of the Unicorn, Inc. ("MOTU") owns both this program and its documentation. Both the program and the documentation are protected under applicable copyright laws. Your right to use the program and the documentation are limited to the terms and conditions described herein.

License

YOU MAY: (a) use the enclosed program on a single computer; (b) physically transfer the program from one computer to another provided that the program is used on only one computer at a time and that you remove any copies of the program from the computer from which the program is being transferred; (c) make copies of the program solely for backup purposes. You must reproduce and include the copyright notice on a label on any backup copy.

YOU MAY NOT: (a) distribute copies of the program or the documentation to others; (b) rent, lease or grant sublicenses or other rights to the program; (c) provide use of the program in a computer service business, network, time-sharing, multiple CPU or multiple user arrangement without the prior written consent of MOTU; (d) translate or otherwise alter the program or related documentation without the prior written consent of MOTU.

Term

Your license to use the program and documentation will automatically terminate if you fail to comply with the terms of this Agreement. If this license is terminated you agree to destroy all copies of the program and documentation.

Limited Warranty

MOTU warrants to the original licensee that the disk(s) on which the program is recorded be free from defects in materials and workmanship under normal use for a period of ninety (90) days from the date of purchase as evidenced by a copy of your receipt. If failure of the disk has resulted from accident, abuse or misapplication of the product, then MOTU shall have no responsibility to replace the disk(s) under this Limited Warranty.

THIS LIMITED WARRANTY AND RIGHT OF REPLACEMENT IS IN LIEU OF, AND YOU HEREBY WAIVE, ANY AND ALL OTHER WARRANTIES, BOTH EXPRESS AND IMPLIED, INCLUDING BUT NOT LIMITED TO WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE. THE LIABILITY OF MOTU PURSUANT TO THIS LIMITED WARRANTY SHALL BE LIMITED TO THE REPLACEMENT OF THE DEFECTIVE DISK(S), AND IN NO EVENT SHALL MOTU BE LIABLE FOR INCIDENTAL OR CONSEQUENTIAL DAMAGES, INCLUDING BUT NOT LIMITED TO LOSS OF USE, LOSS OF PROFITS, LOSS OF DATA OR DATA BEING RENDERED INACCURATE, OR LOSSES SUSTAINED BY THIRD PARTIES EVEN IF MOTU HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES. THIS WARRANTY GIVES YOU SPECIFIC LEGAL RIGHTS WHICH MAY VARY FROM STATE TO STATE. SOME STATES DO NOT ALLOW THE LIMITATION OR EXCLUSION OF LIABILITY FOR CONSEQUENTIAL DAMAGES, SO THE ABOVE LIMITATION MAY NOT APPLY TO YOU.

Update Policy

In order to be eligible to obtain updates of the program, you must complete and return the attached Mark of the Unicorn Purchaser Registration Card to MOTU.

General

This License Agreement shall be governed by the laws of the Commonwealth of Massachusetts and shall inure to the benefit of MOTU, its successors, administrators, heirs and assigns.

Copyright Notice

Limited Warranty on Hardware

Mark of the Unicorn, Inc. and S&S Research ("MOTU/S&S") warrant this equipment against defects in materials and workmanship for a period of NINETY (90) DAYS from the date of original retail purchase. This warranty applies only to hardware products; MOTU software is licensed and warranted pursuant to separate written statements.

If you discover a defect, first write or call Mark of the Unicorn at (617) 576-2760 to obtain a Return Merchandise Authorization Number. No service will be performed on any product returned without prior authorization. MOTU will, at its option, repair or replace the product at no charge to you, provided you return it during the warranty period, with transportation charges prepaid, to Mark of the Unicorn, Inc., 1280 Massachusetts Avenue, MA 02138. You must use the product's original packing material for in shipment, and insure the shipment for the value of the product. Please include your name, address, telephone number, a description of the problem, and the original, dated bill of sale with the returned unit and print the Return Merchandise Authorization Number on the outside of the box below the shipping address.

This warranty does not apply if the equipment has been damaged by accident, abuse, misuse, or misapplication; has been modified without the written permission of MOTU, or if the product serial number has been removed or defaced.

ALL IMPLIED WARRANTIES, INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE, ARE LIMITED IN DURATION TO NINETY (90) DAYS FROM THE DATE OF THE ORIGINAL RETAIL PURCHASE OF THIS PRODUCT.

THE WARRANTY AND REMEDIES SET FORTH ABOVE ARE EXCLUSIVE AND IN LIEU OF ALL OTHERS, ORAL OR WRITTEN, EXPRESS OR IMPLIED. No MOTU/S&S dealer, agent, or employee is authorized to make any modification, extension, or addition to this warranty.

MOTU/S&S ARE NOT RESPONSIBLE FOR SPECIAL, INCIDENTAL, OR CONSEQUENTIAL DAMAGES RESULTING FROM ANY BREACH OF WARRANTY, OR UNDER ANY LEGAL THEORY, INCLUDING LOST PROFITS, DOWNTIME, GOODWILL, DAMAGE OR REPLACEMENT OF EQUIPMENT AND PROPERTY AND COST OF RECOVERING REPROGRAMMING, OR REPRODUCING ANY PROGRAM OR DATA STORED IN OR USED WITH MOTU/S&S PRODUCTS.

Some states do not allow the exclusion or limitation of implied warranties or liability for incidental or consequential damages, so the above limitation or exclusion may not apply to you. This warranty gives you specific legal rights, and you may have other rights which vary from state to state.

MOTU, AudioDesk and Mark of the Unicorn are trademarks of Mark of the Unicorn, Inc.

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

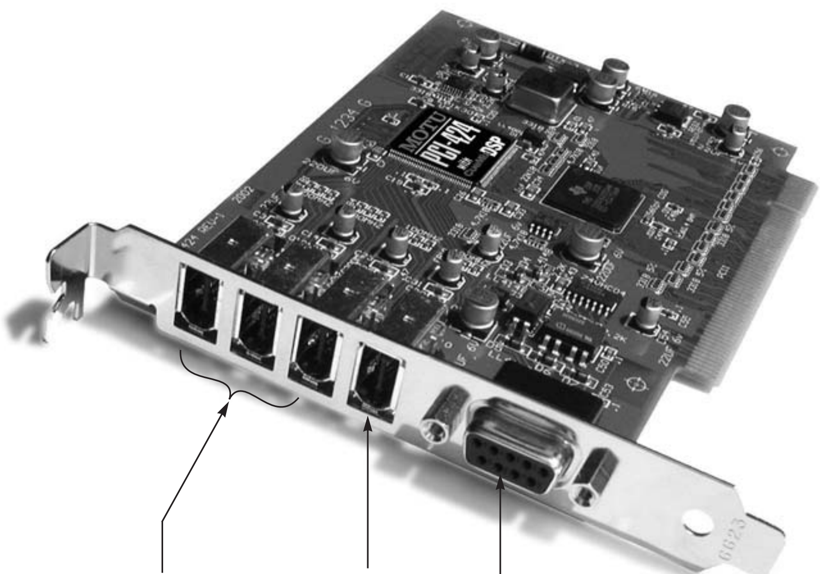
If necessary, you can consult a dealer or experienced radio/television technician for additional assistance.

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.

Contents

Quick Reference: PCI-424 Audio Card



Connect additional MOTU PCI audio interfaces here. You can connect up to four I/Os to a PCI-424 card for a maximum of 96 possible active inputs and outputs.

Connect the first (core system) interface to the PCI-424 card here using the Audio Wire cable included with your core system. If you use a custom cable, it should not be longer than 50 feet.

If you are using the PCI-424 system with one or more ADATs, or any other ADAT Sync-compatible recorder, use this standard ADAT SYNC INPUT to connect the PCI-424 card to the end of your ADAT sync chain. For example, if you have three ADATs, chain the ADATs in the usual fashion (SYNC OUT to SYNC IN, etc.), and then connect the last ADAT's SYNC OUT to this SYNC IN on the PCI-424 card. This connection allows you to make sample-accurate audio transfers between your host audio software (if it supports sample-accurate sync) and the ADATs. If you have a MIDI Timepiece AV or a Digital Timepiece, make it the master of the ADAT SYNC chain so that you can control everything from your host audio software (or your other MIDI Machine Control compatible software).

Quick Reference: 2408mk3 I/O Front Panel

This is a standard quarter-inch stereo headphone jack. Its output matches analog outputs 1 and 2, but the volume knob to the right controls the headphone output only.

This knob controls the volume of the headphone output.

This knob controls the volume of the main outs on the rear panel.

Internally, the 2408mk3 is divided into three banks of I/O: Banks A, B and C. This section of the 2408mk3's front panel shows you the input and output activity for each bank if it has been assigned to either the ADAT, Tascam or SPDIF digital I/O formats. When you are operating the 2408mk3 under computer control, the 2408mk3 always outputs to both ADAT and Tascam, regardless of which one you have chosen in AudioDesk or the PCI-424 configuration software. The top (orange) row of lights show input activity for each channel in the bank, and the bottom (green) lights show output. At 88.2 or 96 kHz, TDIF and ADAT optical support 4 channels per bank.

When the METERS light is illuminated (on the right), this section of LEDs provides metering for 2408mk3's analog input and output activity. Each channel has a 5-segment column of LEDs measuring -42 dB to 0 dB.

When the METERS light is *not* illuminated, The LEDs in this section indicate the CLOCK, SOURCE, and BOUNCE settings in the 2408mk3. When the 2408mk3 is under control of the PCI-424 card (and the host computer), you cannot change the settings using the SELECT and SET buttons, as they are under control of software running on the computer. The only thing you can do in this mode is press the SELECT button to switch between the clock status display and analog metering.

When these three lights are illuminated, they indicate that the 2408mk3 is operating under control of the PCI-424 card. They also indicate that the LEDs in the ANALOG OUT section are displaying the 2408mk3's current clock settings (as you have chosen in the PCI Audio Console software). Press the Select button to toggle the ANALOG OUT section between this clock display and analog output metering.

Clock

When the 2408mk3 is operating as a stand-alone format converter (no 2408mk3-compatible software is running on the computer or the computer is turned off), the CLOCK setting determines what digital clock the 2408mk3 is slaving to. The choices are:

- 44 / 48 / 88 / 96 - These four LEDs indicate the sample rate. If the 2408mk3 is set to slave from an external source, and no clock source is present, these lights blink to show that the 2408mk3 does not currently detect a clock signal. If the 2408mk3 is running under its internal clock, use the SET button to choose the desired sample rate.
- Int (Internal) - Means that the 2408mk3 is running under its own internal clock.
- PCI - This LED means that the 2408mk3 is slaved to the PCI-424 card. This light is not available when the 2408mk3 is in stand-alone mode.
- Dig (Digital input) - This option refers to the digital I/O format being recorded. For example, if you are converting from ADAT to Tascam, and you have selected ADAT as the source, the 2408mk3 will slave to the clock supplied by the ADAT optical digital input.
- Word - the 2408mk3 is slaved to its word clock input.
- Vid (video) - the 2408mk3 is slaved to a video signal received on its VIDEO IN (BNC) connector.
- LTC (Longitudinal Time Code) - the 2408mk3 is slaved to SMPTE time code on an analog input.

Source

When the 2408mk3 is operating as a stand-alone format converter (no 2408mk3-compatible software is running on the computer or the computer is turned off), the Source setting determines which audio format (ADAT, Tascam, Analog or SPDIF) will be the source of the transfer. Use the SELECT button to activate (illuminate) the SOURCE LED. Press the SET button repeatedly to make the desired source setting. Your choices are:

- Analog (all inputs at +4)
- Analog (all inputs at -10)
- Analog (mixed input levels)
- The stereo SPDIF input
- ADAT Bank A
- ADAT Bank B
- ADAT Bank C
- ADAT banks A, B and C
- Tascam (with the same bank choices as shown above for ADAT)

The setting for Analog with mixed input levels uses the input level settings you've specified with the MOTU PCI Audio Console (by clicking the *Interface Options* button).

Bounce

When the 2408mk3 is operating as a stand-alone format converter (no 2408mk3-compatible software is running on the computer or the computer is turned off), the BOUNCE setting lets you shift tracks during a transfer. Use the SELECT button to activate (illuminate) the BOUNCE LED. Press the SET button repeatedly to make the desired bounce setting. Your choices are:

- No shifting (1-2)
- Shift all tracks down by two (3-4)
- Shift all tracks down by four (5-6)
- Shift all tracks down by six (7-8)

Tracks that are shifted lower than track 1 wrap around to track 8 and shift down from there.

The "Swap L to R" option swaps the channels in each pair, in addition to the amount they are shifted. This allows you to copy any source track to any destination track. For example, you can copy track 4 to track 1 by choosing "3-4" and "Swap L to R."

When the 2408mk3 is not under PCI control (no 2408mk3-compatible software is running, or the computer is turned off), the 2408mk3 operates as a stand-alone format converter, and you can use the SELECT and SET buttons to change the CLOCK, SOURCE, and BOUNCE settings.

When the 2408mk3 is resolving to video or SMPTE time code, the LOCK light glows green when lockup has been achieved. The TACH light blinks once per second when the 2408mk3 is successfully reading address (time code) information.

You can use the SET button only when the 2408mk3 is operating as a stand-alone format converter (no 2408mk3-compatible software is running on the computer or the computer is turned off). When the 2408mk3 is in stand-alone operation, press the SET button repeatedly to cycle through the current CLOCK, SOURCE or BOUNCE settings, which control what the 2408mk3 will do as a format converter.

When the 2408mk3 is in stand-alone operation, press the SELECT button repeatedly to illuminate the desired CLOCK, SOURCE or BOUNCE LED. Then press the SET button repeatedly to make the desired setting.

When the 2408mk3 is under control of the computer, press the SELECT button to toggle between the clock display and analog metering in the ANALOG OUT section of the front panel.

Quick Reference: 2408mk3 I/O Rear Panel

These jacks provide stereo SPDIF digital input and output. To use them, choose the SPDIF format for one of the three banks in the PCI-424 configuration window on your computer. Then assign the audio tracks in your software to the SPDIF outputs that appear in its output assignment menus.

These jacks also double as (analog audio) SMPTE time code inputs and outputs. For complete details, see "Syncing to SMPTE time code" on page 41 and chapter 14, "MOTU SMPTE Console" (page 117).

Connect up to three Tascam digital recorders or other TDIF compatible digital I/O devices to these three connectors.

If you want, you can connect both an ADAT optical and a TDIF device to a bank. They will share the same output (regardless of which format you choose on the computer), but you can only record input from one format or the other at a time per bank (as chosen in the PCI Console software).

The 2408mk3's eight analog outputs are gold-plated, balanced +4dB TRS (tip/ring/sleeve) quarter-inch connectors. They have 24-bit, 128x oversampling converters.

Analog outputs 1 and 2 are duplicated on the 2408mk3's main stereo outputs, as well as the front panel stereo headphone jack and the DAT SPDIF output.

Use the word clock input and output for digital transfers with devices that cannot slave to the clock supplied by their digital I/O connection with the 2408mk3. Via software, you can switch the word clock output rate to either double or halve the 2408mk3's system word rate. For example, if the 2408mk3 is running at 96 kHz, it can transmit 48 kHz word clock output.

The WORD IN connector can also serve as a video input, which allows the 2408mk3 to resolve to video or blackburst. For details, see "Syncing to video" on page 42 and chapter 14, "MOTU SMPTE Console" (page 117).

This is an extra SPDIF stereo output jack, which carries a digital copy of the same signal as the main outs and Analog 1-2. You can connect it to whatever you want. For example, you could connect it to a DAT machine to record stereo mixdowns of your 2408mk3 projects.

Connect the 2408mk3 to the PCI-424 card here using the Audio Wire cable provided with your 2408mk3 system.

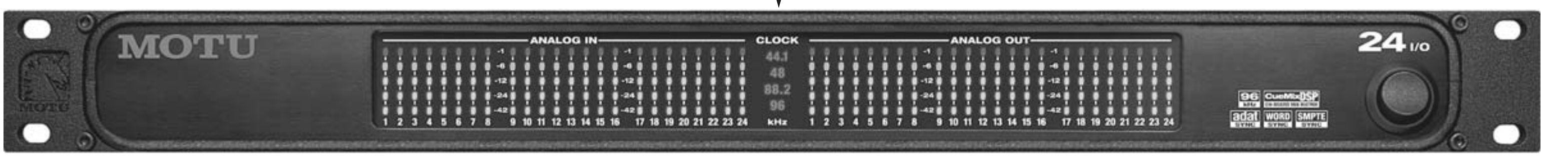
Connect up to three ADATs or other ADAT optical digital I/O devices to these three connectors. Be sure to connect the optical cables "OUT to IN" and "IN to OUT".

The 2408mk3's eight analog inputs are gold-plated, balanced +4dB TRS (tip/ring/sleeve) quarter-inch connectors. They have 24-bit, 64x oversampling converters. Each input pair can be set at +4dB or -10dB via the PCI-424 configuration window on your computer. Click the *Interface Options* button to access the input level settings. Note that you can use one of these inputs for SMPTE time code input, instead of the RCA connectors on the left side of the unit.

These two balanced, quarter-inch jacks serve as the 2408mk3's main outputs. They duplicate the material from channels 1 and 2 of the analog bank. To hear material from them, choose "Analog" as the desired format for one of the 2408mk3's three banks (using the PCI-424 configuration window on your computer), and then assign any tracks you want to hear to Analog channels 1-2. Use the front panel MAIN OUT volume knob to control the level from these outputs.

Quick Reference: 24I/O Front Panel

Indicates the current sample rate. If this LED flashes, it means that you have chosen (via software) an external clock source (such as word clock), but the 24I/O is not successfully receiving it. Check the 24I/O's clock setting in the MOTU PCI Audio Console window, or check your external clock source and its cable connections to the 24I/O.



Five segment metering for the 24 analog inputs.

Five segment metering for the 24 analog outputs.

Quick Reference: 24I/O Rear Panel

The word clock connector can be configured as either an input or an output. When you choose the *24I/O: Word Clock In* setting from the PCI-424 Clock Source menu (see "Clock Source" on page 66), the 24I/O's word clock jack becomes a word clock input. When you choose any other clock setting, it operates as a word clock

The 24I/O's twenty four analog outputs are gold-plated, balanced +4 dB TRS connectors. They have 24-bit, 128x oversampling D/A converters.

The 24I/O's twenty four analog inputs are gold-plated, balanced TRS connectors. They have 24-bit, 128x oversampling A/D converters. From the factory, they are configured for a +4 dB input reference level. You can switch them to -10 dB inputs in banks of eight. For details, see "24I/O interface options" on page 72.

Connect the 24I/O to the PCI-424 card here using the Audio Wire cable provided with your 24I/O system.

Quick Reference: HD192 I/O Front Panel

When the HD192 system has a stable clock (either internal or external), the system's sample rate is illuminated here. If the HD192 is slaving to an external clock, and the external clock is running at either 2 times or one-half the system clock, then both samples rates flash. For example, if the HD192 system clock is running at 96 kHz, but it is slaving to a 48 kHz word clock, then both rates will flash here.

When the HD192 system has no clock signal for some reason, the 'no clock' LED illuminates. Check your cables and clock settings in the software.

The top red 'over' LED lights up when the signal reaches full scale — for even just one sample — and remains illuminated until you clear it in the software. The second 'over' LED below only lights up momentarily so that you can continue to adjust level even after clipping has just occurred.

These LEDs indicate the current clock and sample rate conversion settings for the AES/EBU section of the HD192. All of these settings are made in the MOTU PCI Audio Console (see page 15 for a one-page overview).

19-segment ladder LEDs for the twelve XLR analog inputs.

19-segment ladder LEDs for the twelve XLR analog outputs.

AES/EBU Rate Convert IN

The AES/EBU input can either 1) slave to the HD192 system clock (*Rate convert AES IN* is dark) or 2) sample rate convert the incoming signal (*Rate convert AES IN* illuminates). To enable sample rate conversion, check the *Rate Convert* check box in the Interface Options window in MOTU PCI Audio Console (page 15).

AES/EBU Rate Convert OUT / with external clock

The AES/EBU output can either 1) match the HD192 system clock (*OUT* is dark) or 2) run at different sample rate *OUT* lights up), either under its own clock (*with external clock* is dark) or slaved to the AES input or AES Word In (*with external clock* lights up).

44.1, 48, 88.2 or 96 kHz

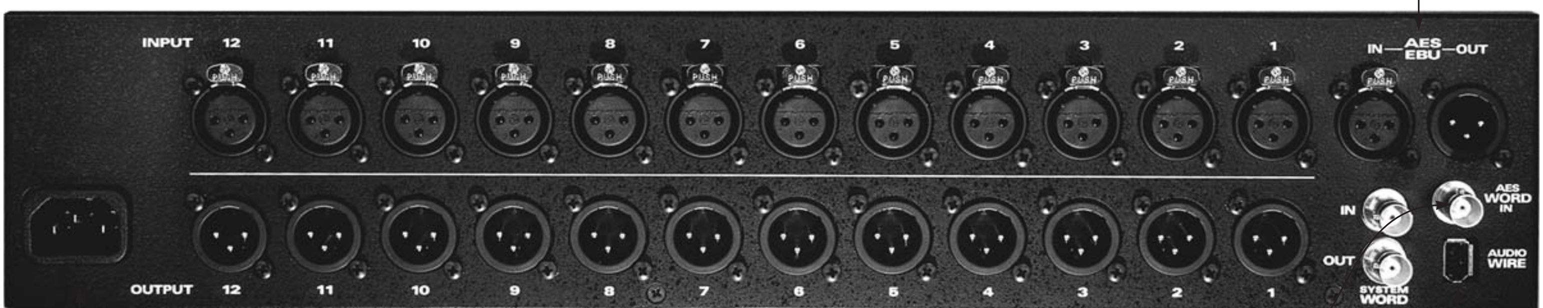
One of these four orange LEDs lights up to indicate the current AES OUT clock rate, even if it matches the current HD192 system clock rate (the large green sample rate above).

Quick Reference: HD192 I/O Rear Panel

These jacks provide stereo AES/EBU digital input and output. For information about configuring the AES/EBU input and output, see "HD192 interface options" on page 73.

When you're recording from the AES/EBU input, either use it as the clock source for the entire system or use its built-in sample-rate converter. For details, see "HD192 AES/EBU" on page 31.

The HD192's twelve analog inputs are balanced, +4dB XLR connectors. They have 24-bit, 128x enhanced multibit A/D converters capable of 44.1, 48, 88.2, 96, 176.4 and 192 kHz recording.



The HD192 power supply is switchable between 115v and 240v operation. It should already be set to the proper voltage for your country, but you can check the setting and adjust it if necessary with the red switch just to the left of this power cord receptacle (on the side of the metal chassis).

The HD192's twelve analog outputs are balanced, +4dB XLR connectors. They have 24-bit, 128x enhanced multibit A/D converters capable of 44.1, 48, 88.2, 176.4 and 192 kHz playback.

'AES Word In' allows you to slave the AES/EBU output to an independent word clock source, separate from the HD192 system itself, which slaves to the 'System Word' input.

Connect the HD192 to the PCI-424 card here using the Audio Wire cable provided with your HD192 system.

Use the word clock input and output for digital transfers with devices that cannot slave to the clock supplied by their digital I/O connection with the HD192.

Quick Reference: Expansion Interfaces

EXPANSION I/O OR CORE SYSTEM?

Read this chapter if you have purchased a MOTU PCI audio interface as an Expansion I/O for a PCI-424 core system. If not, skip to the next chapter.

PACKING LIST FOR AN EXPANSION INTERFACE

Each MOTU Expansion interface ships with:

- One rack-mount audio interface (2408mk3, 24I/O or HD192)
- One 15-foot "Audio Wire" cable
- One CD-ROM with drivers and Setup Wizard
- Power cord
- One 2408mk3 manual and reg card

INSTALLING AN EXPANSION INTERFACE

To connect your expansion interface to a PCI-424 core system, use one of the three available Audio Wire sockets on your core system's PCI-424 card as shown on page 5.

Connect the other end of the Audio Wire cable to the Audio Wire socket on the interface as demonstrated below with the 2408mk3 in Figure 1.

Figure 1: Attaching the Audio Wire cable to an expansion interface.

IMPORTANT NOTE

Always power on your expansion interface when operating your core system. In fact, you should turn on all audio interfaces connected to the PCI-424 card.

COMPLETING THE INSTALLATION

To complete the installation, open the PCI Audio Console to confirm that the PCI-424 card sees the newly installed interface and to configure your multi-interface system. For details, turn to chapter 11, "Expanding Your PCI-424 System" (page 101).

SYNCHRONIZATION AND CLOCK SOURCE

For details about synchronizing an expanded system, see "Synchronizing multiple interfaces" on page 103.

Quick Reference: PCI Audio Console Window

If you have two or more interfaces connected to the PCI-424 card in your computer, use this menu to choose which one you are controlling with the settings in the middle portion of this window.

The 'Clock Source' menu determines the master clock source for your entire PCI-424 system. This is an important setting, as PCI-424 interfaces must be carefully resolved with the clocks in the other digital audio devices connected to them.

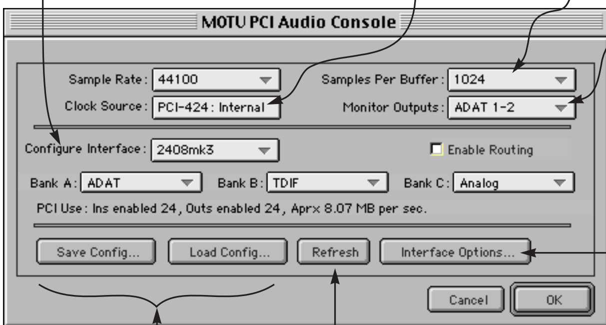
Choosing a smaller setting here reduces the latency you may hear when monitoring live inputs through plug-in effects in your host software. But lower settings also increase the strain on your computer. For details, see "Samples Per Buffer" on page 69.

The 'Monitor Outputs' menu determines which outputs the PCI-424's Sound Manager driver will use.

This section of the window has general settings for the entire PCI-424 system.

This section of the window shows the I/O formats provided by the interface currently chosen in the *Configure Interface* menu.

Provides additional settings for the interface currently chosen in the *Configure Interface* menu. For details, see "Interface options" on page 72.



These buttons let you save and reload the interface settings (in the middle portion of the window).

Refresh checks to make sure that the computer knows about all connected MOTU PCI interfaces.

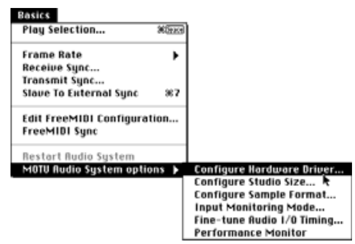
How to open this window

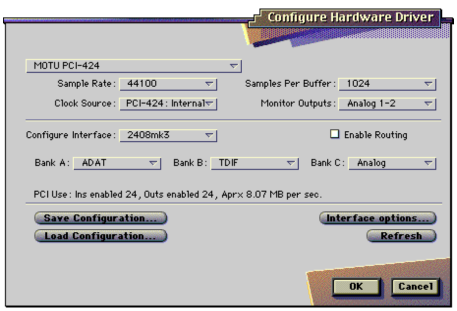
The MOTU PCI Audio Console Window gives you complete control over the settings in your PCI-424 hard disk recording system. There are several ways to access the PCI Audio Console window. But the window is the same, regardless of how you access it.

- From the Mac OS desktop, run MOTU PCI Audio Console (the stand-alone applet for the 2408mk3)

MOTU PCI Audio Console

- From within AudioDesk™ or Digital Performer™, go to the Basics menu and choose *MOTU Audio System options> Configure Hardware Driver*

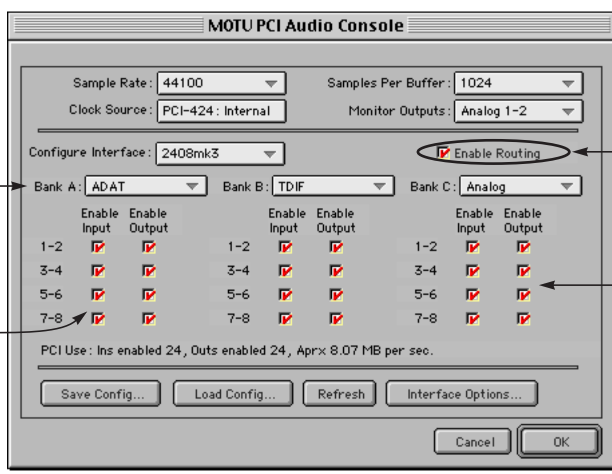




The stand-alone Console window, above, is the same as the "Configure Hardware Driver" dialog found in AudioDesk and Digital Performer (in their Basics menu).

In this example, a 2408mk3 interface is configured for ADAT optical I/O for Bank A, Tascam TDIF I/O for bank B, and Analog for Bank C.

The 'Enable Input' and 'Enable output' check boxes refer to input and output to and from the computer. If checked, inputs and outputs will be available in the menus of AudioDesk, Digital Performer, or other host audio applications that support the 2408mk3.



Check 'Enable Routing' to expand the window as shown. This view lets you enable or disable individual inputs and outputs.

At low sample rates (44.1 or 48 kHz), digital I/O banks provide 8 channels. At high sample rates (88.2 or 96 kHz), digital I/O banks provide 4 channels (two stereo pairs).

CHAPTER 1 About the PCI-424 System

OVERVIEW

The PCI-424 core system is a computer-based hard disk recording system for Mac OS and Windows that offers 24 simultaneous inputs and outputs per Audio Wire cable, expandable to 96 inputs/outputs. A core system consists of a PCI card connected to a standard 19-inch, single-space, rack-mountable audio interface.

The system includes AudioDesk™, full-featured audio workstation software for Mac OS that supports both 16-bit and 24-bit recording at any standard sample rate up to 192 kHz.

For Windows, a WDM driver is included for compatibility with audio applications that support standard multi-channel WDM and Wave drivers.

Also included are Macintosh and Windows ASIO drivers for multi-channel operation with Steinberg Cubase and other ASIO-compatible software.

A note about Mac OS X

This manual covers the operation of the PCI-424 system under Mac OS 9. Visit www.motu.com for the latest information about using the PCI-424 system with Mac OS X.

THE PCI-424 CARD

A PCI-424 core system ships with a single PCI audio card called the PCI-424. The card features a custom processor, a powerful DSP chip, four 6-pin Audio Wire connectors and an ADAT SYNC IN connector.

High definition audio

The PCI-424 system can operate at the following high sample rates: 44.1, 48, 88.2, 96, 176.4 and 192 kHz. All MOTU PCI-424 audio interfaces support rates up to 96 kHz. In addition, the HD192 interface offers analog recording and playback at 176.4 or 192 kHz.

Audio Wire expansion

The PCI-424 card provides four Audio Wire jacks, which are used to attach MOTU PCI audio interfaces to the card installed in the computer. Each Audio Wire is capable of transmitting 24 simultaneous channels of 96 kHz digital audio input and output or 12 simultaneous channels of 192 kHz input and output.

Up to four MOTU PCI audio interfaces can be connected to a single PCI-424 card for a maximum of 96 possible input and output connections. All MOTU PCI audio interfaces (including the 2408mk3, 24I/O and HD192) can also be "mixed and matched" on the PCI-424 card, along with legacy PCI-324 interfaces, such as the 2408, 2408mII, 1296, 1224, 24i and 308.

Audio Wire carries a proprietary communication protocol between the card and the external I/O to handle the extremely low latencies required by the system. The heart of the PCI-424 card is a custom-programmed VLSI chip capable of simultaneously processing all 96 inputs and outputs (192 channels total) at samples rates up to 96 KHz. At 176.4 or 192 kHz, this chip can process 48 simultaneous channels of input and output.

The custom chip handles all of the system's I/O processing, freeing up the host computer's processing bandwidth for real-time DSP effects and hard disk I/O.

CueMIx DSP

The PCI–424 card features CueMix DSP™, a flexible DSP-driven mixing and monitoring matrix that provides the same near-zero monitoring latency as today's latest digital mixers. CueMix DSP™ allows you to connect keyboards, synth modules, drum machines, and even effects processors and then monitor these live inputs with no audible delay and no processor drain on the host computer's CPU. The CueMix DSP engine resides on the PCI–424 card, so it works across all interfaces connected to the card. The included CueMix Console software provides an on-screen mixer that gives you hands-on control of your monitor mix, regardless of what audio software you prefer to use. Digital Performer users have the additional option of controlling CueMix DSP directly within Digital Performer's mixing environment. CueMix DSP completely eliminates the buffer latency associated with monitoring on host-based recording systems.

Sample-accurate synchronization

The PCI-424 card's standard 9-pin ADAT SYNC IN connector provides sample-accurate synchronization with all ADATs or other ADAT Sync-compatible devices connected to the system. For example, if you digitally transfer a single track of material from the ADAT via light pipe into the 2408mk3's Macintosh workstation software, and then transfer the track back to the ADAT, it will be recorded exactly at its original location, down to the sample.

Video and SMPTE time code synchronization

The entire PCI-424 system, including all connected interfaces, can resolve directly to SMPTE time being received on any analog input in the system. In addition, the 2408mk3 interface provides a BNC video input for resolving the entire system directly to video.

THE 2408MK3 INTERFACE

24 simultaneous inputs and outputs

The 2408mk3 I/O is a single-space, rack mountable chassis with gold-plated analog and digital audio connectors on its rear panel and status LEDs on the front. The rear panel has seven banks of 8-channel I/O at 44.1 or 48 kHz in the following formats:

- One bank of 8 balanced (+4 dB) analog quarter-inch (TRS) inputs and outputs.
- Three Tascam DA-88 'TDIF' 8-channel digital I/O connectors.
- Three Alesis ADAT optical 'light pipe' 8-channel digital I/O connectors.

Three banks of 8-channel I/O at 44.1/48 kHz

Internally, the 2408mk3 has three 8-channel I/O busses (A, B and C) for a total of 24 simultaneous inputs and outputs at 44.1 or 48 kHz. Using the included console software (available for both Mac and PC), you can freely choose any I/O format for each bank. For example, you can choose analog for Bank A, ADAT optical for Bank B, and Tascam TDIF for Bank C. Or you can choose ADAT optical for all three banks.

Three banks of 4-channel digital I/O at 88.2/96

At 88.2 or 96 kHz, the 2408mk3 interface offers 4 channels of I/O per bank for the ADAT optical and Tascam TDIF formats, along with 8 channels of 96 kHz analog input and output. Therefore, the 2408mk3 interface offers a maximum of 16 channels of input and output at 96 kHz (2 banks of 4-channel digital I/O plus 8 channels of analog I/O). The maximum number of digital inputs and outputs at 96 kHz is 12 (3 banks of 4-channel I/O).

Analog bank

The analog inputs are equipped with 24-bit, 96 kHz 64x oversampling A/D converters. The analog outputs have 24-bit, 96 kHz 128x oversampling D/A converters.

The entire analog section of the circuit board inside the 2408mk3 is physically isolated from the rest of the board to help ensure quiet analog performance.

Each of the 2408mk3's eight analog stereo input pairs can be switched between -10 dB and +4 dB operating levels to support a wide variety of input sources, including synths, samplers and other equipment.

Main Outs

For main stereo output, analog outputs 1 and 2 are duplicated on the rear panel as a stereo pair of balanced, +4dB quarter-inch TRS jacks. A dedicated knob on the front panel gives you volume control of the main outs.

SPDIF

The 2408mk3 rear panel has three SPDIF stereo pairs. The SPDIF output on the right is dedicated to duplicating the stereo main out so that you can, for example, conveniently record a stereo mix to your DAT deck without swapping cables with other SPDIF devices. The other two SPDIF connectors serve as independent stereo inputs/outputs. They can be used with any SPDIF compatible device, such as an effects processor or other device.

Word Clock

BNC Word clock connectors (in and out) are provided for synchronization with standard word clock devices. Via software, the word output can be made to either double or halve the 2408mk3's system word rate. For example, if the 2408mk3 is running at 96 kHz, it can generate 48 kHz word clock output.

Video sync

The BNC input connector can be switched via software to become a dedicated video input, allow you to slave your PCI-424 system to NTSC or PAL/SECAM video (or blackburst) without a dedicated synchronizer. The PCI-424 card provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy.

SMPTE time code

The RCA jacks for SPDIF input/output can be switched via software to become a dedicated SMPTE time code (LTC) input and output, allowing you to slave your PCI-424 system to time code and/or generate time code. If you prefer to use them for SPDIF digital I/O, you can use any analog input to receive SMPTE time code. Any active channel, digital or analog, can be chosen as a SMPTE time code output.

Level Meters

The front panel of the 2408mk3 I/O displays several banks of status LEDs. On the left are three banks of eight LEDs (A, B and C) that show audio signal on the 2408mk3's three ADAT/TDIF digital I/O buses. On the right are eight vertical, five-segment LEDs that show input level from the eight analog inputs measured from -40 dB to 0 dB. A similar bank of dedicated five-segment LEDs is provided for the analog outputs. These LEDs also provide status information, such as the system clock sample rate and other settings.

Stand-alone Format Conversion

Finally, several additional LEDs, along with accompanying set/select buttons, allow you to access the 2408mk3 I/O's stand-alone format conversion features. With these controls, you can bounce any I/O format to any other format without a computer.

Headphone output

The 2408mk3 front panel includes a quarter-inch stereo headphone output jack and volume knob. The headphone output matches the main stereo outs, which have their own volume knob.

THE 24I/O INTERFACE

24 analog 24-bit/96 kHz inputs and outputs

The 24I/O is a single-space, rack mountable chassis with gold-plated analog audio connectors on its rear panel and level meters on the front. The rear panel has 24 gold-plated, balanced +4dB TRS analog inputs and outputs with 24-bit A/D converters. All inputs and outputs can be accessed simultaneously. Internally, the 24I/O has a 24-bit data path to and from the computer so that all audio data is carried to/from the computer in 24 bits.

Converters

All analog-to-digital and digital-to-analog inputs on the 24I/O are equipped with 24-bit, 128x oversampling, extremely high-quality, latest-generation converters.

Word clock in and out

BNC Word clock connector can be programmed via software for either input or output for synchronization with standard word clock devices.

Level Meters

The front panel of the 24I/O displays 48 level meters, arranged in three banks of 8 channels. For each channel, there is a five-segment meter that measures from -42 dB to 0 dB.

The clock section in the middle of the front panel indicates the current sample rate.

THE HD192 INTERFACE

High definition audio recording

The HD192 provides the very best A/D and D/A conversion available. It can operate at any standard sample rate from 44.1 kHz to 192 kHz.

12 analog inputs and 12 outputs

The HD192 I/O is a two-space, rack mountable chassis with XLR audio connectors on its rear panel and level meters on the front. The rear panel has four banks of I/O in the following formats:

- One bank of 12 XLR inputs
- One bank of 12 XLR outputs
- One pair of AES/EBU stereo digital I/O connectors

All analog inputs and outputs can be accessed simultaneously.

AES/EBU digital I/O

The HD192 rear panel includes an AES/EBU stereo digital input and output. Both of these connectors are capable of handling 24-bit digital audio at sample rates up to 96 kHz. The AES/EBU input and output each has its own independent sample rate converter for real-time conversion between any two sample rates between 40 and 100 kHz.

System word clock in and out

BNC 'System' Word clock connectors (in and out) are provided for synchronizing the HD192's main system clock with other standard word clock devices.

AES word clock in

A BNC Word clock input connector is provided for independently synchronizing the AES/EBU output of the HD192 to a standard input clock.

Level Meters

The front panel of the HD192 I/O displays several banks of status LEDs.

On the right are two banks of meters for the twelve analog inputs and twelve analog outputs. For each channel, there is a 19-segment meter that measures from -42 dB to 0 dB, along with two red 'over' lights. The top LED remains illuminated after clipping occurs until it is cleared via software (either manually or via a configurable time-out period). The red LED below it only lights momentarily when clipping.

The *Clock* section on the left displays the current system clock setting (44.1, 48, 88.2, 96, 176.4, 192 or *no clock*), as well as the AES/EBU clock setting if sample rate conversion is taking place independently of the system clock.

16-BIT AND 24-BIT RECORDING

The PCI-424 system handles all data with a 24-bit signal path, regardless of the I/O format. Using AudioDesk™, the PCI-424 system's Macintosh workstation software (included), you can record and play back 16-bit or 24-bit audio files at 44.1, 48, 88.2 or 96 KHz via any of the system's analog or digital inputs and outputs. The HD192 also offers recording at 176.4 or 192 kHz. On Windows, 24-bit audio files can be recorded with any compatible host application that supports 24-bit recording.

AUDIODESK

AudioDesk is a full-featured, 24-bit audio workstation software package for Macintosh included with each PCI-424 core system. AudioDesk provides multi-channel waveform editing, automated virtual mixing, graphic editing of ramp automation, real-time effects plug-ins with 32-bit floating point processing, crossfades, support for many third-party audio plug-ins (in the MOTU Audio System and Adobe Premiere formats), background processing of file-based operations, sample-accurate editing and placement of audio, and more.

DIGITAL PERFORMER

The PCI-424 system is fully integrated with MOTU's award-winning Digital Performer audio sequencer software package.

OTHER AUDIO SOFTWARE

The PCI-424 system ships with a standard WDM driver that allows you to record, edit, play back and mix your projects using your favorite WDM- and Wave-compatible Windows software.

The PCI-424 system also ships with standard Mac OS audio drivers.

The PCI-424 also includes a Macintosh and Windows ASIO driver for multi-channel compatibility with Steinberg Cubase and other ASIO-compatible software.

A COMPUTER-BASED SYSTEM

Regardless of what software you use with the PCI-424 system, the host computer determines the number of tracks the software can record and play simultaneously, as well as the amount of real-time effects processing you can apply to your mix. A faster computer with more RAM and faster hard drives will allow more simultaneous tracks and real-time effects than a slower computer with less RAM and slower hard drives. Standard third-party system acceleration products can also help you achieve higher track counts.

CHAPTER 2 Packing List and Macintosh System Requirements

PACKING LIST

A MOTU PCI-424 core system ships with the items listed below. If any of these items are not present in your core system box when you first open it, please immediately contact your dealer.

- One rack-mountable audio interface (2408mk3, 24I/O or HD192)
- One PCI-424 audio card
- One 15-foot "Audio Wire" cable
- Power cord
- One PCI-424 Mac/Windows "flip book" manual
- One AudioDesk Manual (for Mac OS only)
- One cross-platform CD-ROM
- Product registration card

MACINTOSH SYSTEM REQUIREMENTS

The PCI-424 system requires the following Macintosh system:

- A G3/500 or G4 Power Macintosh or faster
- At least 128 Mb (megabytes) of RAM (512 Mb or more is recommended)
- One available PCI slot
- Mac OS version 9 or later
- A large hard drive (preferably at least 20 GB)

PLEASE REGISTER TODAY!

Please send in the registration card included with your PCI-424 system. As a registered user, you will be eligible to receive on-line technical support email and announcements about product enhancements as soon as they become available. Only registered users receive these special update notices, so please, complete and mail this registration card!

There is also an AudioDesk software registration card found at the beginning of your AudioDesk manual. Please be sure to fill out and return this card as well, so that you will be eligible to receive on-line technical support email and announcements about AudioDesk software enhancements as soon as they become available.

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

CHAPTER 3 Installing the PCI-424 Hardware

OVERVIEW

TRY THE SETUP WIZARD

The PCI-424 software installer provides an easy-to-use Setup Wizard. This interactive software tutorial will help you figure out the best way to connect all your gear to the PCI-424 card and connected audio interfaces. This chapter covers important general concepts regarding connections and synchronization, but the Setup Wizard asks you specific questions about your gear and then makes specific recommendations for you based on your answers. To use the Wizard, just run the software installer on the 2408mk3 CD, and then look for it on your hard drive when the installation is done.

INSTALL THE PCI-424 AUDIO CARD

1 Switch off and unplug your computer.

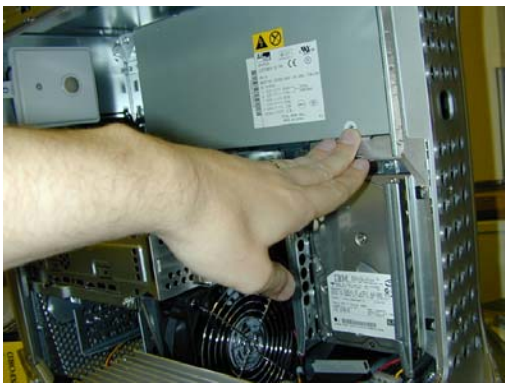
☛ Failure to do so may result in serious shock or injury.

2 Open your computer.

3 Find an available PCI slot.

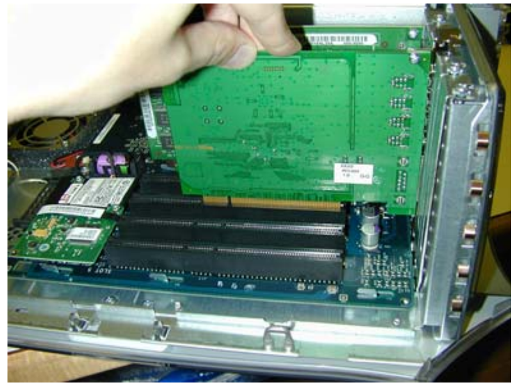
4 Remove the slot cover, if necessary.

5 Before removing the PCI-424 card from it's anti-static bag, touch the power supply inside your computer to discharge any static electricity that may have built up on you.



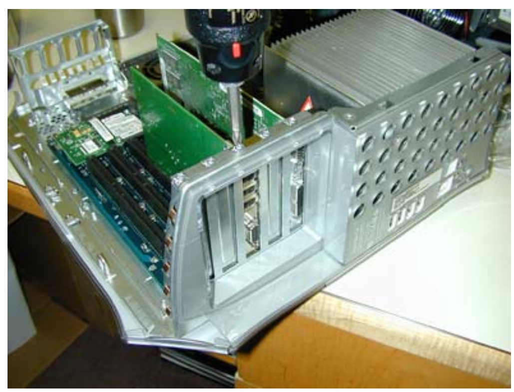
6 Remove the PCI-424 card from its anti-static bag.

7 Gently but firmly insert the card into any available PCI slot.



8 Secure the bulkhead of the PCI-424 card to the computer chassis with the bolt from the slot cover.

☛ We strongly recommend securing the PCI-424 card in this manner. Doing so allows you to ensure secure connections to the card later on in the installation.



9 Place the cover back on your computer.

10 Reconnect the power cord to the computer before proceeding.

CONNECT THE AUDIO INTERFACE

1 Plug one end of the Audio Wire cable (included) into the Audio Wire socket next to the 9-Pin ADAT Sync connector on the PCI-424 card as shown below in Figure 3-1.

2 Plug the other end of the Audio Wire cable into the audio interface as shown below in Figure 3-1.

Step 1

Connect the Audio Wire to the socket closest to the ADAT SYNC port.

Step 2

2408mk3 interface

24I/O interface

HD192 interface

Figure 3-1: Connecting the audio interface to the PCI-424 audio card.

2408MK3 INPUT & OUTPUT CONNECTIONS

The 2408mk3 audio interface has the following input and output connectors:

- 8 balanced, +4 dB quarter-inch analog in/out
- 2 balanced, +4 dB quarter-inch TRS main out
- 1 pair of RCA SPDIF stereo in and out
- 1 extra RCA SPDIF stereo out
- 3 banks of Alesis ADAT optical digital in and out
- 3 banks of Tascam TDIF digital in and out

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

Internally, the 2408mk3 has three separate banks of 8-channel input/output. You can choose any format you want for each bank, and you can freely switch between them at any time. Therefore, you don't have to be too concerned about where you plug things in, with the exceptions noted below.

SPDIF can only be used on Bank C. Therefore, if you would like to use another I/O format at the same time as SPDIF, connect it to Bank A or B.

Reminder: ADAT optical goes OUT to IN and IN to OUT, like MIDI.

It doesn't matter which bank you connect ADAT or TDIF devices to, unless you want to be using them simultaneously with other formats. For example, if you have one ADAT, plus a DAT deck in your studio, and you want to be able to use them both at the same time, don't connect the ADAT to Bank C (since SPDIF can only be used on that bank).

Here's another example, if you have one ADAT and one Tascam recorder, and you want to use them independently at the same time, connect them to different banks.

On the other hand, you can connect an ADAT optical device and Tascam TDIF device to the same bank, if you like. For input, you'll only be able to record from one format or the other at one time (on that bank). But on output, the 2408mk3 actually plays back audio on both formats at the same time, regardless of which one is currently active. This lets you send the same audio material to both formats (but not different audio).

The main outputs, as well as the DAT SPDIF output and the headphone output on the front panel, match the output from Analog outputs 1-2. If none of the 2408mk3's three internal banks are set to the Analog format, the 2408mk3's analog output bank will duplicate one of the three banks. The MOTU PCI Audio Console Window has an *Interface Options* setting that lets you choose which bank. For details, see "Interface options" on page 72.

Here are some example connections:

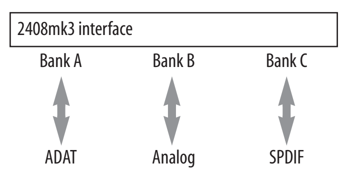


Figure 3-3: A recommended setup for one ADAT optical device, a DAT deck, and miscellaneous analog devices.

Figure 3-2: You can connect up to three ADAT optical devices, three Tascam TDIF devices, eight analog and two SPDIF devices to the 2408mk3.

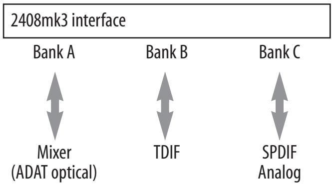


Figure 3-4: A recommended setup for a digital mixer with ADAT optical I/O, one TDIF device, a DAT deck and analog devices.

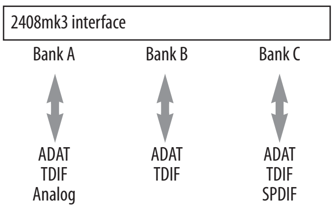


Figure 3-5: A recommended setup for a full blown setup of three ADAT optical devices, three TDIF devices, analog and SPDIF.

Connecting analog devices with -10 dB signals

From the factory, the 2408mk3 provides +4 dB of headroom for analog input signals. However, if you're connecting a synthesizer, drum machine, sampler, or other device with a -10 dB analog input signal, you can switch it's analog input pair to -10 dB. For details see "2408mk3 interface options" on page 72.

Mixing live inputs with CueMix DSP

The PCI-424 system is ideal for computer-based studios where mixing is done entirely in the computer and for more advanced installations built around a digital mixer of any size. For the computer-based studio, the PCI–424 card features CueMix DSP™, a flexible DSP-driven mixing and monitoring matrix that provides the same near-zero monitoring latency as today's latest digital mixers.

CueMix DSP gives you a great number of choices for mixing and monitoring inputs in your studio. For example, you could connect the analog output of a synth module to a pair of 2408mk3 (or other interface) analog inputs, and then bus that signal via CueMix DSP to a pair of analog outputs connected to an effects processor – say a rack-mount reverb unit. The output from the effects processor could then be fed back into a second pair of 2408mk3 inputs (just like an aux return) and then routed to audio software running on the host computer, as well as the 2408mk3 main outputs.

The result is that you can apply hardware reverb to the live synth input, listen to it on your studio monitors with no audible delay while also recording it into your workstation software (either wet or dry), also with no delay. The performance of this signal path is the same as an send/return loop on a conventional digital mixer. CueMix DSP completely eliminates the buffer latency associated with monitoring live inputs on host-based recording systems, and it places no processor drain on your host computer.

If you have a fast computer, CueMix DSP can, of course, be combined with signals monitored through host-based effects as well. A fast computer allows you to drop the host buffer settings low enough to greatly reduce – and even eliminate – audible buffer latency. The combination of CueMix DSP and host-based monitoring with effects processing provides a flexible, powerful system.

For more information, see chapter 12, "Reducing Monitoring Latency" (page 105).

24I/O INPUT AND OUTPUT CONNECTIONS

Here are a few things to keep in mind when making connections to the 24I/O inputs and outputs.

Connect them all, if you like

All of the 24i's inputs and outputs can be active simultaneously.

Avoid adaptors

If you don't have the right cable, you'll avoid headaches later on — and ensure the best possible audio quality — by taking the time to obtain the correct cable.

Connecting analog devices with -10 dB signals

From the factory, the 24I/O provides +4 dB of headroom for analog input signals. However, if you're connecting synthesizers, drum machines, samplers, or other devices with a -10 dB analog input signal, you can switch the 24I/O inputs to -10 dB in banks of eight inputs. For example, you might configure inputs 1 through 16 as inputs from a console at +4 dB, while configuring inputs 17-24 at -10 dB for MIDI instruments. For details see "24I/O interface options" on page 72.

Mixing live inputs with CueMix DSP

As you plan the I/O routing for your 24I/O interface, refer to "Mixing live inputs with CueMix DSP" on page 29. CueMix DSP™ is a flexible DSP-driven mixing and monitoring matrix that provides the same near-zero monitoring latency as today's latest digital mixers, and it will help you determine the mixing and monitoring scheme for your studio.

Figure 3-6: You can connect up to 24 analog inputs and outputs to the 24I/O.

HD192 INPUT AND OUTPUT CONNECTIONS

Here are a few things to keep in mind when making connections to the HD192 inputs and outputs.

Connect them all, if you like

All of the HD192's inputs and outputs can be active simultaneously.

+4 dB analog inputs/outputs

The HD192's analog connectors are calibrated at +4 dB, so if you are plugging in a microphone, you'll need a mic preamp of some kind (or a connection to a mixer with a mic pre).

Mixing live inputs with CueMix DSP

As you plan the I/O routing for your HD192 interface, refer to "Mixing live inputs with CueMix DSP" on page 29. CueMix DSP™ is a flexible DSP-driven mixing and monitoring matrix that provides the same near-zero monitoring latency as today's latest digital mixers, and it will help you determine the mixing and monitoring scheme for your studio.

HD192 AES/EBU

If you would like to transfer stereo audio digitally between the HD192 and another device that has AES/EBU I/O, connect it to the HD192's AES/EBU jacks with balanced, AES/EBU grade audio cables. Connecting the HD192's AES/EBU jacks to RCA SPDIF connectors via adaptors is not recommended. The HD192's AES/EBU connectors support 24-bit digital audio at any sample rate up to 100 kHz.

Internally, the HD192 is "hard-wired" to provide a maximum of 12 independent inputs and outputs. Both the analog section and AES/EBU section share these I/O resources, with AES/EBU taking up one pair of the 12 virtual ins and outs. For example, in the MOTU PCI Audio Console, you could use channels 1 through 10 for analog input and channels 11-12 for AES/EBU input. AES/EBU completely takes over the inputs, entirely disabling the corresponding analog inputs. For output, however, AES/EBU mirrors its corresponding pair of analog outputs, so you can send the same stereo mix to both analog and AES/EBU simultaneously.

Therefore, if you plan to use AES/EBU at the same time as the analog inputs/outputs of the HD192, think about which analog pair you are least likely to need while transferring audio via AES/EBU. Plug those analog ins/outs into the sockets on the rear panel that correspond to the channel numbers you'd like to use for AES/EBU. (The channels numbers you use for AES/EBU don't matter. Use whatever you like.)

AES/EBU clock and sample rate conversion

The HD192 AES/EBU section is equipped with real-time sample rate converters on both input and output. In addition, the AES/EBU section has its own clock crystal, allowing it to run at a completely independent sample rate from the HD192 system clock. Together, these features provide a great deal of flexibility in making digital transfers. For example, you can:

Figure 3-7: You can connect up to 12 analog inputs and outputs, plus stereo AES/EBU, to the HD192.

■ Transfer digital audio into or out of the HD192 at a sample rate that is completely different than the HD192 system clock rate.

■ Transfer digital audio into the HD192 without the need for any external synchronization arrangements.

■ Receive AES/EBU input at one sample rate, run the HD192 system clock at a second sample rate, and send AES/EBU output at yet a third sample rate.

Rate conversion does not add any appreciable noise to the audio signal (under -120 dB).

Digital I/O without sample rate conversion

Without sample rate conversion, when you transfer digital audio between two devices, their audio clocks must be in phase with one another — or *phase-locked,* as discussed in "Choosing a digital audio clock master" on page 35.

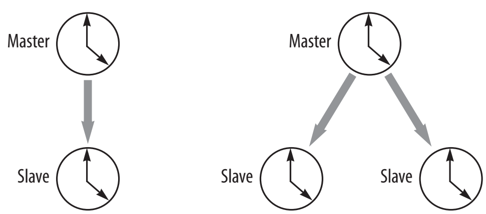


Figure 3-8: With the HD192's sample rate conversion turned off, you need to keep it phased-locked with the other AES/EBU device by choosing a clock master.

Digital I/O with sample rate conversion

With sample rate conversion (SRC), an extra level of master/slave clocking is added to the equation, as demonstrated below in Figure 3-9, which shows the clocking going on when you transfer digital audio from the HD192 (AES/EBU OUT) to a DAT deck (AES/EBU IN) using SRC. Notice that with SRC, the DAT deck is not slaved to the HD192's system clock. Instead, their clocks are running completely independently of one another. But also notice that the DAT deck must still slave to the sample-rate-converted output from the HD192 for a clean digital audio transfer (unless it has its own rate converter on its AES/EBU input).

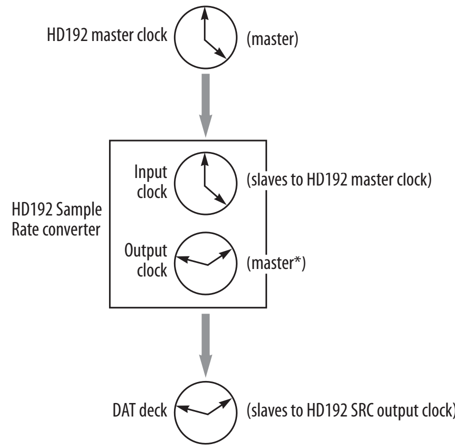


*Figure 3-9: Clock relationships when sending audio from the HD192 to a DAT deck using sample rate conversion. The DAT deck needs to be slaving to its AES/EBU input. *Note: the HD192 AES/EBU output can actually be clocked from a number of different sources. In this example, it is running under its own crystal. For details about other possible clock sources, see "HD192 interface options" on page 73.*

System clock, AES clock & rate convert settings

When you are setting up AES/EBU input and output with the HD192, pay careful attention to the following settings in the MOTU PCI Audio Console window:

■ System clock

■ AES input options (Interface options button)

■ AES output options (Interface options button)

Clocking scenarios for AES/EBU input

There are three possible clocking scenarios for the HD192 AES/EBU input:

1. Simple transfer (slave the HD192 system clock to the AES/EBU input signal — no sample rate conversion).

2. Sample rate convert the AES/EBU input.

3. Use word clock to resolve the HD192 system clock and the other AES/EBU device with each other.

These three AES/EBU input scenarios are summarized below.

	Scenario 1	Scenario 2	Scenario 3
Description	Simple transfer	Rate convert	Use word clock
HD192 system clock setting	HD192: AES/EBU	Any setting except HD192: AES/EBU	HD192: Word Clock
AES/EBU input rate convert	unchecked (off)	checked (on)	unchecked (off)
Required HD192 cable connections	AES/EBU IN	AES/EBU IN	AES/EBU IN and Word Clock IN
Are the devices continuously resolved?	Yes	No	Yes
Is the signal being sample rate converted?	No	Yes	No
Example application	Simple digital transfer into the HD192 from DAT deck or digital mixer.	Transfer from digital mixer running at a different sample rate.	Both the HD192 and other AES/EBU device are slaved to 'house" word clock.

Some example scenarios are demonstrated below.

Simple AES/EBU input transfer (no rate convert)

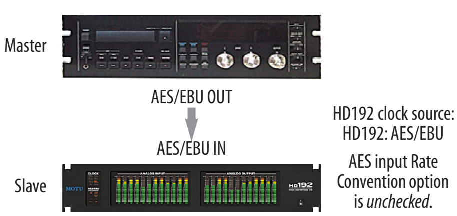


Figure 3-10: Slaving the HD192 to an AES/EBU device. For the HD192's clock source, choose 'HD192: AES/EBU'.

AES/EBU input with rate conversion

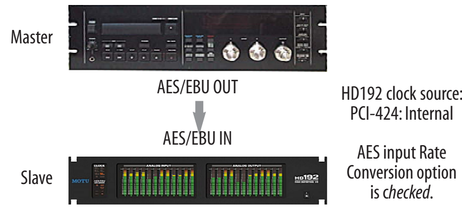


Figure 3-11: Rate-converting AES/EBU input.

AES/EBU input with word clock

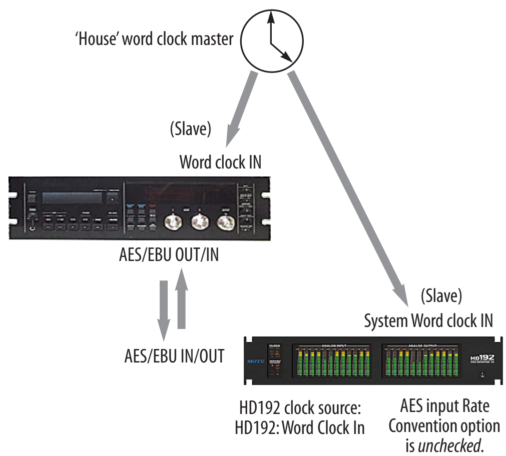


Figure 3-12: In this scenario, the HD192 and other AES/EBU device are both resolved to one another via a third master word clock source.

Clocking scenarios for AES/EBU output

The HD192 AES/EBU output is equipped with an independent sample rate converter. In addition, it can be clocked from one of several possible sources, including its own crystal and separate 'AES Word Input'. These options, shown below in Figure 3-13, are briefly summarized in the following sections. For further details, see "HD192 interface options" on page 73.

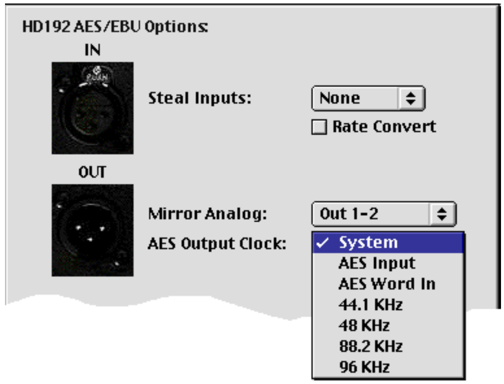


Figure 3-13: Click the 'Interface Options' button in the MOTU PCI Audio Console window to access the AES/EBU output clock options.

System

To make the AES/EBU output sample rate match the System sample rate, choose *System*. No additional connections are necessary. And no sample rate conversion occurs when this setting is chosen.

AES Input

To make the AES/EBU output sample rate match the sample rate currently being received by the HD192's AES/EBU input, choose *AES Input*. This setting requires a connection to the HD192's AES/EBU input from a device that is transmitting an AES/EBU clock signal.

☛ Be careful when both the HD192's AES/EBU input and output are connected to the same external device: this option is likely to create a clock loop.

AES Word In

If you have an external word clock source, and you would like the HD192 AES/EBU output to resolve to it, connect the word clock source to the 'HD192 AES Word In' as shown below in Figure 3-14. Then, choose *AES Word In* from the *AES Output Clock* menu (Figure 3-13 above).

Figure 3-14: Resolving the AES/EBU output to its own independent word clock.

44.1 / 48 / 88.2 / 96 kHz

Choose one of these sample rates when the desired AES/EBU output rate needs to be different than the HD192 system clock rate. When you choose one of these rates, an additional option appears called *Fixed Frequency*, which lets you resolve the AES/EU output rate to the HD192 system clock. In either case, sample rate conversion occurs if the sample rate you've chosen is different than the HD192 system clock rate. For further details about this option, see "HD192 interface options" on page 73.

MAKE SYNC CONNECTIONS

Synchronization between the PCI-424 system and the devices connected to it is critical, even if you don't plan to synchronize your PCI-424 system with an outside time reference such as SMPTE time code. While there are dozens of ways to synchronize the system, the next few pages discuss common recommended setups.

Do you need to synchronize the PCI-424?

If you will be using only analog inputs and outputs, and you have no plans to synchronize your PCI-424 system to SMPTE time code, you don't need to make any sync connections. You can skip this section and proceed to chapter 4, "Installing the PCI-424 Macintosh Software" (page 53). After you install the 2408mk3 software, you'll open the PCI Audio Console Window and set the *Clock Source* setting to *Internal* as shown below in Figure 3-15. For details, see chapter 6, "MOTU PCI Audio Console" (page 65).

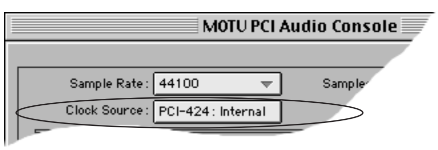


Figure 3-15: You can run the PCI-424 system under its own internal clock when it has no digital audio connections and you are not synchronizing the PCI-424 system to SMPTE time code or video.

Situations that require synchronization

There are three general cases in which you will need to synchronize the PCI-424 system with other devices:

- The 2408mk3 or other interface is connected to other digital audio devices, and their digital audio clocks need to be *phase-locked* (as shown in Figure 3-16).
- You need to resolve (synchronize) the PCI-424 system to SMPTE time code and/or video.
- Both of the above.

Synchronization is essential for digital I/O

Synchronization is critical in any audio system, but it is especially important when you are transferring audio between digital audio devices. Your success in using the PCI-424 system depends almost entirely on proper synchronization. The following sections guide you through several recommended scenarios.

Choosing a digital audio clock master

When you transfer digital audio between two devices, their audio clocks must be in phase with one another — or *phase-locked*. Otherwise, you'll hear clicks, pops, and distortion in the audio — or perhaps no audio at all.

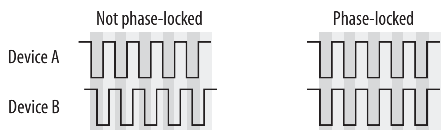


Figure 3-16: When transferring audio, two devices must have phased-locked audio clocks to prevent clicks, pops or other artifacts.

There are two ways to achieve phase lock: slave one device to the other, or slave both devices to a third master clock. If you have three or more digital audio devices, you need to slave them all to a single master audio clock.

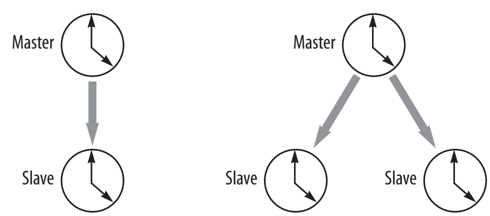


Figure 3-17: To maintain phase-lock between the PCI-424 system and other digital audio devices connected to it, choose a clock master.

Also remember that audio phase lock can be achieved independently of time code (location). For example, one device can be the time code master while another is the audio clock master. But only one device can be the audio clock master. If you set things up with this rule in mind, you'll have trouble-free audio transfers.

DO YOU NEED A SYNCHRONIZER?

Whether or not you'll need a synchronizer depends on your gear and what you will be doing with your PCI-424 system. The following pages give you specific information about common sync scenarios. At least one of them will likely apply to you. Here are some general considerations to help you figure out if you need (or want) a synchronizer for you PCI-424 system.

You don't need a synchronizer if...

As explained earlier, the PCI-424 system's digital audio clock must be phase-locked (synchronized) with other connected digital audio devices to achieve clean digital transfers between them. Can this be accomplished without an additional digital audio synchronizer? It depends on the nature of the other devices, and what you want to do with them. You don't need a synchronizer if the device has a way of locking itself directly to the PCI-424 system's clock (via ADAT lightpipe or word clock, for example), AND if the device carries no sense of location in time. A digital mixer is a good example: it can slave to its ADAT lightpipe connection from a 2408mk3, and it has no sense of time; it just passes audio through for mixing.

A stand-alone digital recorder, on the other hand, does have a sense of location in time, either via SMPTE time code or via its own sample address. For example, if you want to fly tracks back and forth between your computer and an Alesis hard disk recorder while maintaining the audio's position in time, the ADAT Sync port on the PCI-424 card lets you do so without a separate synchronizer — and with sample-accurate precision, as long as you're using AudioDesk, Digital Performer, or other sample-accurate software. Just connect the PCI-424 card directly to the Alesis recorder (or other ADAT SYNC-compatible device) as discussed in "Sample-accurate ADAT sync with no synchronizer" on page 39. But if you also want transport control over the entire rig (including the hard disk recorder) from your audio software, you'll need a MIDI Machine Control-compatible synchronizer such as MOTU's MIDI Timepiece AV, as discussed in "Sample-accurate sync" on page 37. If you are simply using the stand-alone recorder as a way to capture live tracks that you then transfer in one pass into the computer, no synchronizer is required because the tracks will remain in perfect phase lock with each other as you transfer them together. In this scenario, you can simply slave the stand-alone recorder to the optical output from a 2408mk3 as explained in "Syncing ADAT 'lightpipe' devices" on page 46.

Transport control from your computer

If you have stand-alone digital recorders connected to a 2408mk3, and they support ADAT Sync or Tascam Sync, your audio software — if it supports MIDI Machine Control (MMC) — allows you to control the transports of everything from your computer. Most advanced audio programs support MMC. To do this, you'll also need an MMC-compatible ADAT or Tascam synchronizer, such as a MOTU MIDI Timepiece AV or Digital Timepiece. Synchronizers like these allow you to play, stop, rewind and locate all of your tape decks using the transport controls in the audio software. If your audio software supports sample-accurate sync (like Digital Performer and AudioDesk), you can do all of this with sample-accurate precision. The following pages show you how to achieve MMC control, where possible.

Continuous sync to video & SMPTE time code

The PCI-424 system can synchronize directly to video and/or SMPTE time code. If your audio software supports sample-accurate sync (like Digital Performer and AudioDesk), it can also resolve to video and/or time code via the PCI-424. If your software does not support sample-accurate sync, you need a dedicated synchronizer, as illustrated on the following pages.

SAMPLE-ACCURATE SYNC

Your PCI-424 system provides you with the most advanced, accurate synchronization possible with Alesis and Tascam modular digital tape decks and hard disk recorders—or any device that supports the sample-accurate ADAT and Tascam sync formats. Figure 3-18 below shows a few best-case scenarios. Below is a brief explanation of the benefits you achieve with these setups.

Sample accurate locating

With *sample accurate locating*, when you transfer audio between AudioDesk (or any other sample-accurate host software) and a sample-accurate recorder, the audio will not drift in time — even by as little as one sample. This is the tightest possible synchronization between digital audio devices. The timing in your audio will not be affected in any way by the process of transferring it between the PCI-424 system and the recorder.

Is your audio software sample-accurate?

Sample-accurate locating is only possible with software that supports this feature, such as AudioDesk or Digital Performer. For third-party software, sample-accurate performance (if it's supported) is achieved through the PCI-424's ASIO Version 2 driver.

Transport control from your computer

If you have a MIDI Timepiece AV, Digital Timepiece, Alesis BRC, or any ADAT synchronizer that also supports MIDI Machine Control (MMC), you can play, stop, rewind and locate all of your Alesis recorders (or other ADAT SYNC-compatible devices) using the transport controls in the audio software running on your computer. This includes cueing features like markers, position bars, playback wipers, time rulers, etc. If you have Tascam recorders, a Digital Timepiece (or other MMC-compatible Tascam synchronizer) provides this feature.

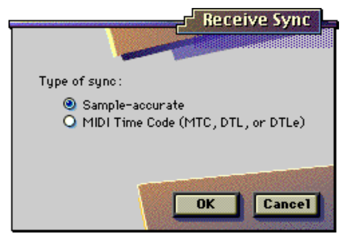


Figure 3-19: AudioDesk and Digital Performer support sample-accurate transfers with Alesis and Tascam modular digital tape decks and hard disk recorders.

Sync format	Software	Synchronizer	Sample accurate locating	Transport control from computer	Continuous sync to SMPTE / MTC
ADAT	AudioDesk, Cubase, or Digital Performer	MIDI Timepiece AV	Yes	Yes	Yes
ADAT	AudioDesk, Cubase, or Digital Performer	Digital Timepiece	Yes	Yes	Yes
ADAT	AudioDesk, Cubase, or Digital Performer	BRC (or any MMC capable ADAT synchronizer)	Yes	Yes	Yes
Tascam (and/or ADATs)	AudioDesk, Cubase, or Digital Performer	Digital Timepiece	Yes	Yes	Yes

Figure 3-18: These recommended combinations of hardware and software offer the tightest sync possible between the PCI-424 system and digital audio recorders in the form of sample-accurate locating between the software and the hardware. Sample-accurate locating is possible with Alesis recorders even without a MIDI Timepiece AV or Digital Timepiece, although you give up transport control from the computer.

SAMPLE-ACCURATE ADAT SYNC

This page shows an ideal setup for using the PCI-424 and a 2408mk3 interface with one or more ADATs, Alesis hard disk recorders or any ADAT SYNC-compatible devices. Connect the PCI-424 card to the end of the ADAT chain and make the software settings as shown below in Figure 3-20. This setup is also ideal if you have both Alesis and Tascam decks. If so, connect your Tascam equipment to the Digital Timepiece as directed in the Digital Timepiece manual.

☛ For ADAT optical digital mixers, see “Syncing ADAT ‘lightpipe’ devices” on page 46.

Use this setup if you have:
✓ ADATs, Alesis hard disk recorders or any ADAT SYNC compatible device(s).
✓ A MOTU Digital Timepiece, MIDI Timepiece AV or other ADAT synchronizer.
✓ Host software that supports sample-accurate sync.

This setup provides:
✓ Sample-accurate locating between all ADAT SYNC-compatible devices, the 2408mk3 and your software (AudioDesk, Digital Performer or other sample-accurate software).
✓ With a Digital Timepiece, this setup provides sample-accurate locating across all devices: ADAT, Tascam and the 2408mk3.
✓ Transport control of everything from the computer, OR continuous sync to SMPTE time code and other sync sources (the other source is the transport master in this case).

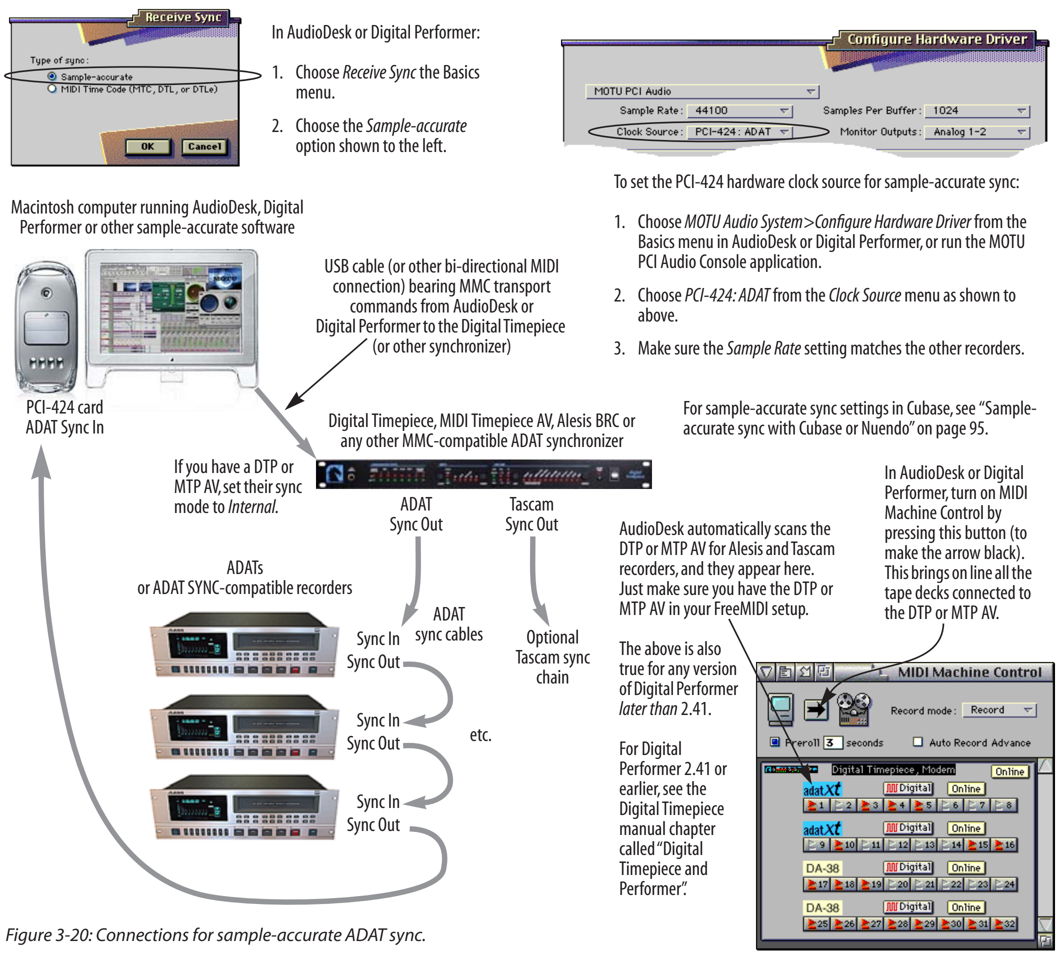
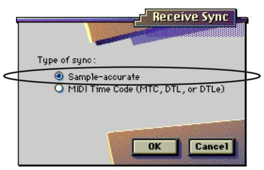


Figure 3-20: Connections for sample-accurate ADAT sync.

SAMPLE-ACCURATE ADAT SYNC WITH NO SYNCHRONIZER

Even if you don't have an ADAT synchronizer, you can achieve sample-accurate sync between ADAT SYNC-compatible devices, a 2408mk3, and AudioDesk, Digital Performer or other sample-accurate software. Just connect the PCI-424 card to the end of the ADAT sync chain as shown below. But without the synchronizer, you don't get transport control from your computer, nor can you slave the system to external SMPTE time code. Instead, you have to play, stop, rewind and cue the system from the transports on your ADAT-compatible recorder.

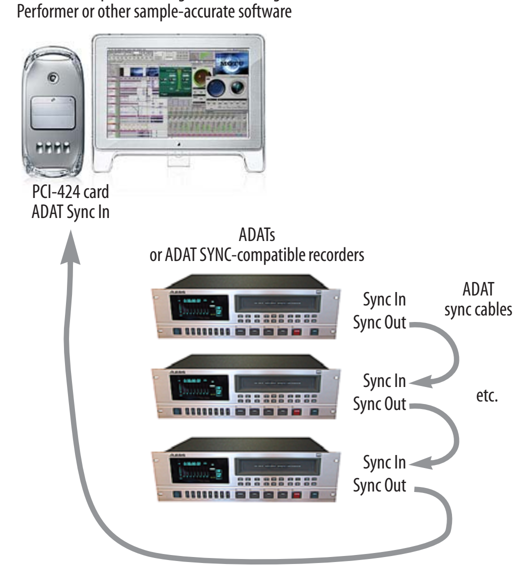
Use this setup if you have:	
✓	ADATs, Alesis hard disk recorders or any ADAT SYNC compatible device(s).
✗	No ADAT synchronizer.
✓	Host software that supports sample-accurate sync.
This setup provides:	
✓	Sample-accurate locating between all ADAT SYNC-compatible devices, the 2408mk3 and your software (AudioDesk, Digital Performer or other sample-accurate software).
✗	No sync with Tascam recorders.
✗	No transport control of everything from the computer.
✗	No sync to SMPTE time code or other sync sources.

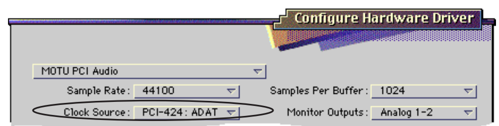


In AudioDesk or Digital Performer:

1. Choose *Receive Sync* from the Basics menu.
2. Choose the *Sample-accurate* option shown to the left.

Macintosh computer running AudioDesk, Digital Performer or other sample-accurate software



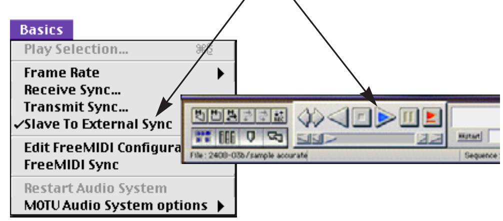


To set the PCI-424 hardware clock source for sample-accurate sync:

1. Choose *MOTU Audio System>Configure Hardware Driver* from the Basics menu in AudioDesk or Digital Performer, or run the MOTU PCI Audio Console application.
2. Choose *PCI-424: ADAT* from the *Clock Source* menu as shown to above.
3. Make sure the *Sample Rate* setting matches the tape decks.

In AudioDesk or Digital Performer:

1. Make sure that *Slave to External Sync is* checked in the Basics menu.
2. Click the play or record button. The software will then wait for you to start your ADAT.
3. Press the Play button on the front panel of your ADAT to initiate playback or recording.



For sample-accurate sync settings in Cubase, see "Sample-accurate sync with Cubase or Nuendo" on page 95.

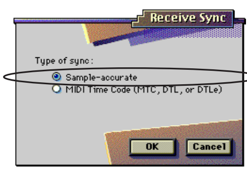
Figure 3-21: Sample-accurate sync between AudioDesk or Digital Performer and one or more ADAT SYNC-compatible devices—without an ADAT synchronizer.

SAMPLE-ACCURATE TASCAM SYNC

This page shows how to set up sample-accurate locating (the most accurate synchronization possible) between a 2408mk3 interface and Tascam digital recorders. This setup requires a Digital Timepiece (or other MMC-compatible Tascam synchronizer). If you also have Alesis recorders, see page 38. If you have a DA-88/SY-88 or DA-98 and you want to operate in offset mode from the time code track, see "Syncing to a DA-88/98 time code track" on page 45.

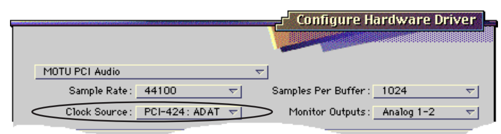
Use this setup if you have:
✓ Tascam DA-series digital tape decks or hard disk recorders.
✓ A Digital Timepiece or other Tascam synchronizer.
✓ Host software that supports sample-accurate sync.

This setup provides:
✓ Sample-accurate locating between all Tascam sync-compatible devices, the 2408mk3 and your software (AudioDesk, Digital Performer or other sample-accurate software).
✓ If you also have ADATs, this setup provides sample-accurate locating across all devices: ADAT, Tascam and the 2408mk3.
✓ Transport control of everything from the computer, OR continuous sync to SMPTE time code and other sync sources (the other source is the transport master in this case).



In AudioDesk or Digital Performer:

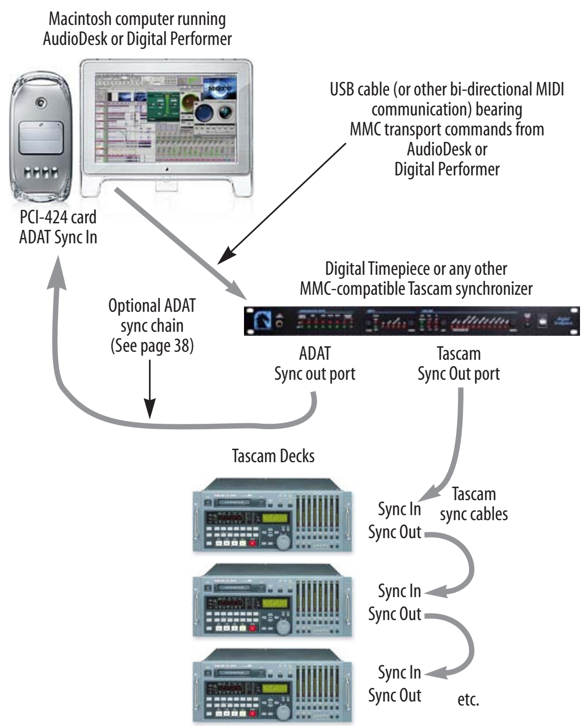
1. Choose *Receive Sync* from the Basics menu.
2. Choose the *Sample-accurate* option shown to the left.



To set the PCI-424 hardware clock source for sample-accurate sync with Tascam:

1. Choose *MOTU Audio System>Configure Hardware Driver* from the Basics menu in AudioDesk or Digital Performer, or run the MOTU PCI Audio Console application.
2. Choose *PCI-424: ADAT* from the *Clock Source* menu as shown to above.
3. Make sure the *Sample Rate* setting matches the tape decks.

For sample-accurate sync settings in Cubase, see "Sample-accurate sync with Cubase or Nuendo" on page 95.



Set the Digital Timepiece sync mode to *Internal*.

In AudioDesk or Digital Performer, turn on MIDI Machine Control by pressing this button (to make the arrow black). This brings "on line" all the tape decks connected to the DTP or MTP AV.

AudioDesk automatically scans the DTP or MTP AV for Alesis and Tascam recorders, and they appear here. Just make sure you have the DTP or MTP AV in your FreeMIDI setup.

The above is also true for any version of Digital Performer *later than* 2.41.

For Digital Performer 2.41 or earlier, see the Digital Timepiece manual chapter called "Digital Timepiece and

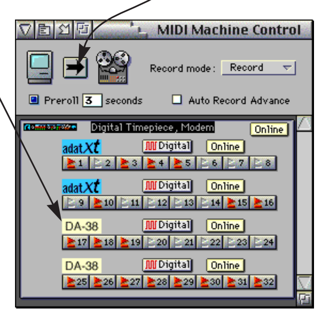


Figure 3-22: Connections for sample-accurate sync between one or more Tascam recorders and a 2408mk3 interface.

SYNCING TO SMPTE TIME CODE

The PCI-424 system can resolve directly to SMPTE time code from any analog input on any audio interface. It can also generate time code and word clock, under its own clock or while slaving to time code. Therefore, the system can act both as an audio interface and digital audio synchronizer, to which you can slave other digital audio devices. You can use the PCI-424 system to slave your audio software to SMPTE as well, as long as your software supports sample-accurate sync, which is the means by which the software follows the PCI-424. The accuracy may not be sample-accurate, but in most cases it will be pretty close. If you would like to resolve to video and SMPTE, see the next page.

Use this setup if you have:
✓ A SMPTE time code source, such as a multitrack tape deck.
✓ A PCI-424 core system by itself, OR an expanded PCI-424 system that includes other MOTU PCI interfaces and maybe one other slaved device (such as a digital mixer).
✓ Host software that supports sample-accurate sync.
This setup provides:
✗ No sample-accurate locating.
✓ Continuous sync to SMPTE time code.
✓ Sub-frame timing accuracy.
✓ Transport control from the SMPTE time code source.

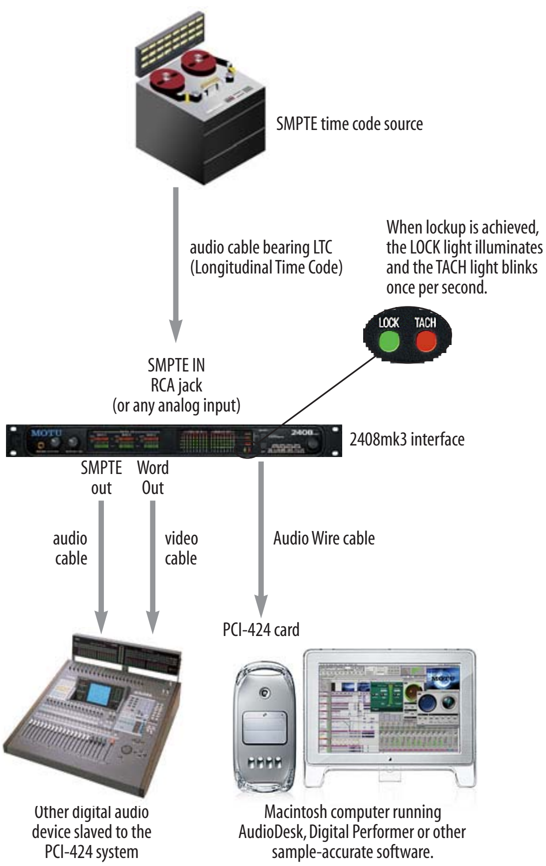
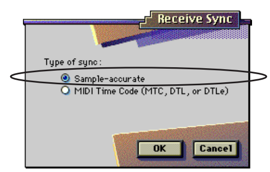
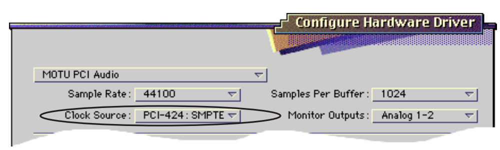


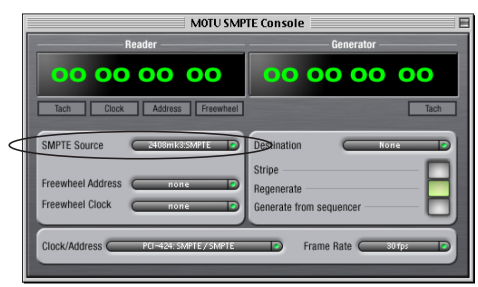
Figure 3-23: Connections for synchronizing the PCI-424 system directly to SMPTE time code.



In AudioDesk or Digital Performer, choose *Receive Sync* from the Basics menu. Choose the *Sample-accurate* option. For sample-accurate sync settings in Cubase, see "Sample-accurate sync with Cubase or Nuendo" on page 95.



Choose *PCI-424: SMPTE* as the clock source in AudioDesk, Digital Performer, or the MOTU PCI Audio Console application. This setting can also be made in the MOTU SMPTE Console (shown below).



Launch the MOTU SMPTE Console and specify the *SMPTE Source*, which is the analog input receiving the SMPTE time code. Also, confirm that the *Clock Source* is *PCI-424: SMPTE/SMPTE*. For details about the other settings, see chapter 14, "MOTU SMPTE Console" (page 117).

SYNCING TO VIDEO

The 2408mk3 interface can continuously resolve directly to video, while at the same time referencing SMPTE time code, with no additional synchronization device required. When you choose video as the clock source (as shown below), the Word Clock BNC connector switches to a video input. Both NTSC and PAL/SECAM rates are supported. The MOTU SMPTE Console software provides options for freewheeling both time code and video time base. To resolve your audio software to video as well, it must support sample-accurate sync. For details, see “Is your audio software sample-accurate?” on page 37.

Use this setup if you have:
✓ A video deck or other video source, with SMPTE time code.
✓ A 2408mk3 core system by itself, OR a small 2408mk3 system that includes other MOTU PCI interfaces and maybe one other device (such as a digital mixer) slaved to the 2408mk3.
✓ Host software that supports sample-accurate sync.
This setup provides:
✗ No sample-accurate locating.
✓ Continuous sync to SMPTE time code.
✓ Sub-frame timing accuracy.
✓ Transport control from the SMPTE time code source.

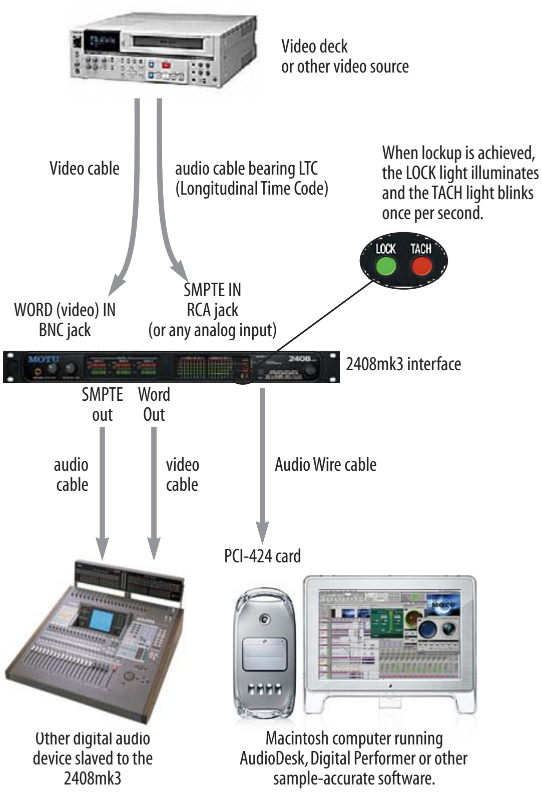
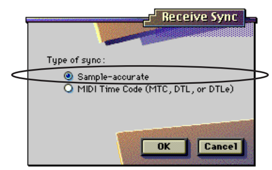
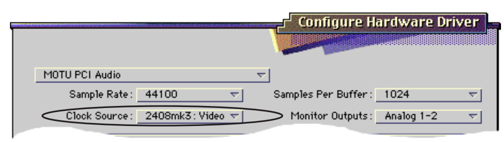


Figure 3-24: Connections for synchronizing a 2408mk3-based PCI-424 system directly to video.

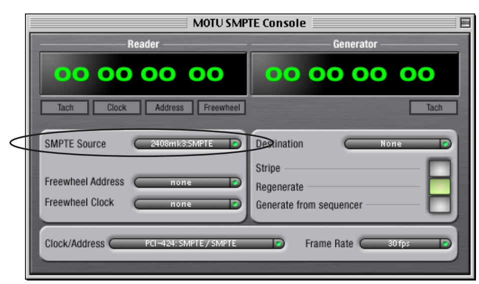


In AudioDesk or Digital Performer, choose *Receive Sync* from the Basics menu. Choose the *Sample-accurate* option.

For sample-accurate sync settings in Cubase, see “Sample-accurate sync with Cubase or Nuendo” on page 95.



Choose *PCI-424: Video* as the clock source in AudioDesk, Digital Performer, or the MOTU PCI Audio Console application. This setting can also be made in the MOTU SMPTE Console (shown below).



Launch the MOTU SMPTE Console and specify the *SMPTE Source*, which is the analog input receiving the SMPTE time code. Also, confirm that the *Clock Source* is *PCI-424: SMPTE/SMPTE*. For details about the other settings, see chapter 14, “MOTU SMPTE Console” (page 117).

SYNCING TO VIDEO AND/OR SMPTE TIME CODE USING A SYNCHRONIZER

If your host audio software does not support the PCI-424's on-board SMPTE sync features (because your software does not support sample-accurate sync), you need a universal synchronizer, such as a MOTU MIDI Timepiece AV or Digital Timepiece. These dedicated sync boxes can read video and SMPTE time code and then convert it into word clock and MIDI Time Code (MTC). The word clock goes to your MOTU PCI audio interface (such as a 2408mk3) to resolve the audio hardware, and MIDI Time Code is fed to your host audio software, which locks to it, as shown below in Figure 3-25.

Use this setup if you have:
✓ Video and/or a SMPTE time code source.
✓ A PCI-424 system of any size.
✓ A Digital Timepiece, MIDI Timepiece AV or other universal synchronizer.
✓ Host software that does not support sample-accurate sync (although you can use this setup even if it does).

This setup provides:
✗ No sample-accurate locating.
✓ Continuous sync to SMPTE time code.
✓ Sub-frame timing accuracy.
✓ Transport control from the SMPTE time code source.

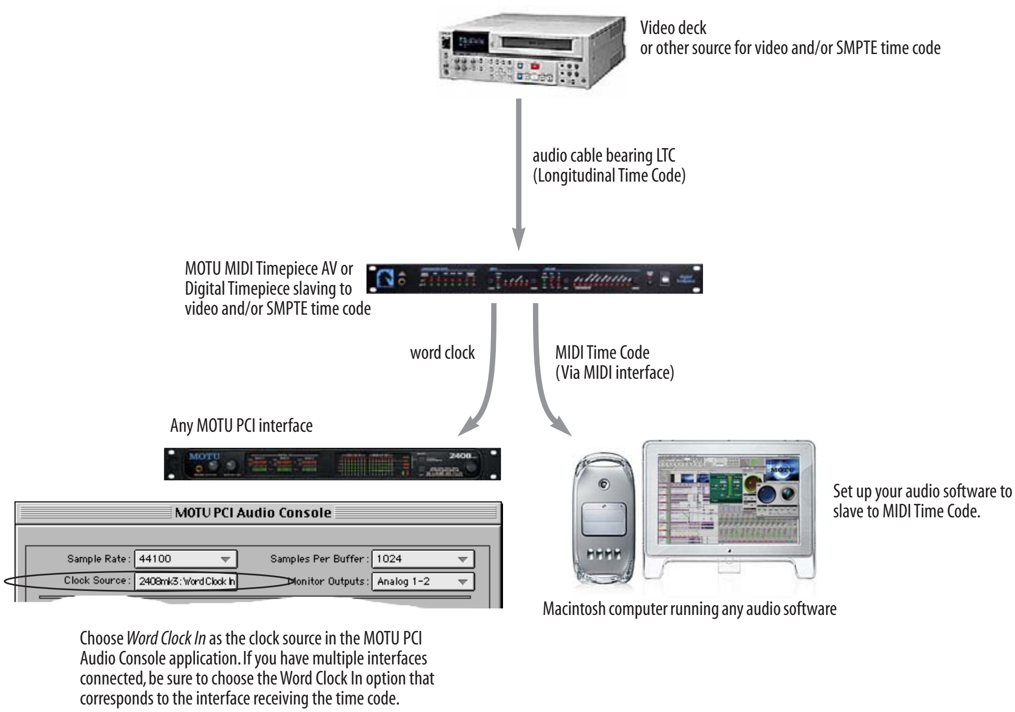


Figure 3-25: If your host audio software does not support sample-accurate sync and the PCI-424's built-in SMPTE sync features, use a universal synchronizer such as the MOTU MIDI Timepiece AV or Digital Timepiece.

SYNCING TO ADAT OR TASCAM DEVICES USING SMPTE TIME CODE

The 2408mk3's built-in SMPTE time code sync features described on page 41 and page 42 are ideal for small PCI-424 systems. But if your audio software doesn't support sample-accurate sync, or if you have an elaborate setup that perhaps involves multiple Alesis and/or Tascam digital recorders, then a dedicated synchronizer such as the Digital Timepiece can serve as the "sync hub" for your digital studio. The Digital Timepiece becomes the clock master over the PCI-424 system, your audio software, and other slaved devices so that they remain phased-locked with each other and share a unified time code address. This setup is also the one to use for syncing your PCI-424 system to time code or video if your audio software does not support sample-accurate sync.

Use this setup if you have:

✓ ADATs, Alesis hard disk recorders or any ADAT SYNC compatible device(s),

AND/OR

Tascam DA-series digital tape decks or hard disk recorders.

✓ A MOTU Digital Timepiece, MIDI Timepiece AV or other ADAT synchronizer.

✓ Host software that does not support sample-accurate sync (although you can use this setup even if it does).

Note: for transport control from the computer, your software must support MIDI Machine Control (MMC).

This setup provides:

✗ No sample-accurate locating.

✓ Sub-frame timing accuracy.

✓ Transport control of everything from the computer, OR continuous sync to SMPTE time code and other sync sources (the other source is the transport master in this case).

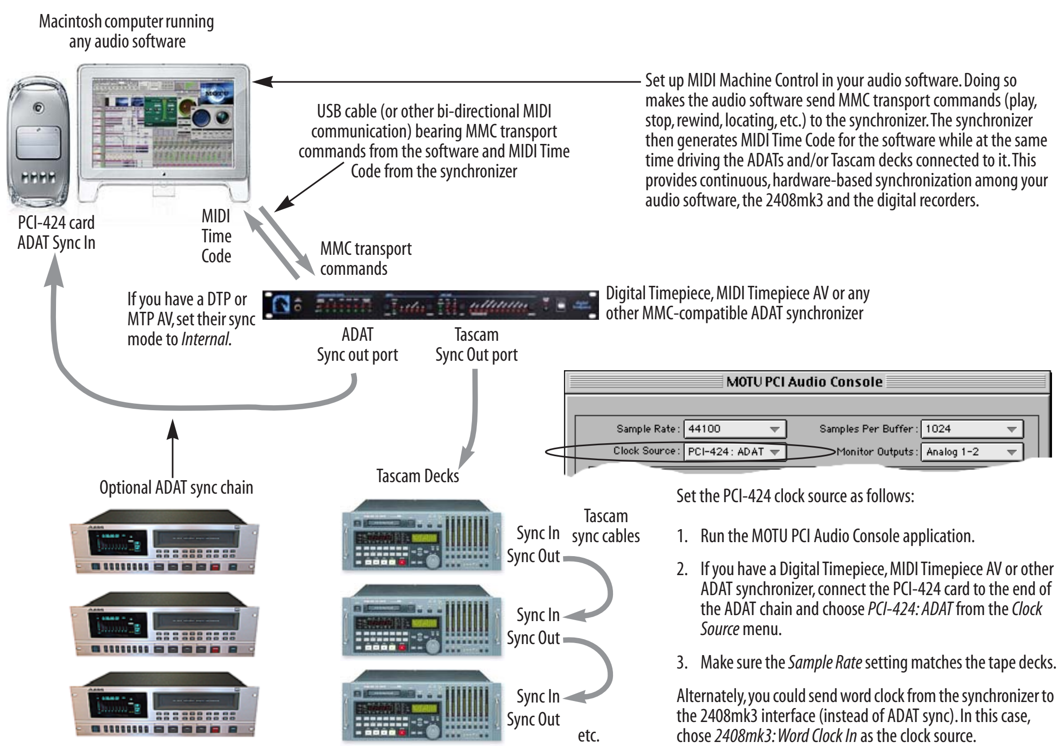


Figure 3-26: An ADAT synchronizer like the Digital Timepiece or MIDI Timepiece AV gives you the best possible synchronization setup for a 2408mk3 with audio software and ADATs. A Digital Timepiece is ideal for Tascam decks (or both ADAT and Tascam).

SYNCING TO A DA-88/98 TIME CODE TRACK

If you have a Tascam DA-88 equipped with an SY-88 card, or a DA-98, and you would like to slave your entire system to its time code track, the DA-88/98 serves as the SMPTE time code master, while the PCI-424 system serves as the word clock master over the DA-88/98. In this setup, you feed MIDI Time Code (MTC) to the computer from the DA-88/98. To establish word clock phase-lock between the DA-88/98, you can either feed word clock from a MOTU PCI audio interface to the DA-88/98 or the opposite (feed word clock from the DA-88/98 to the MOTU interface). Or you can slave them both to a third word clock master, such as a MOTU Digital Timepiece.

Use this setup if you have:

✓ Tascam DA-88 with an SY-88 sync card, a Tascam DA-98 or any other Tascam deck with a time code track.

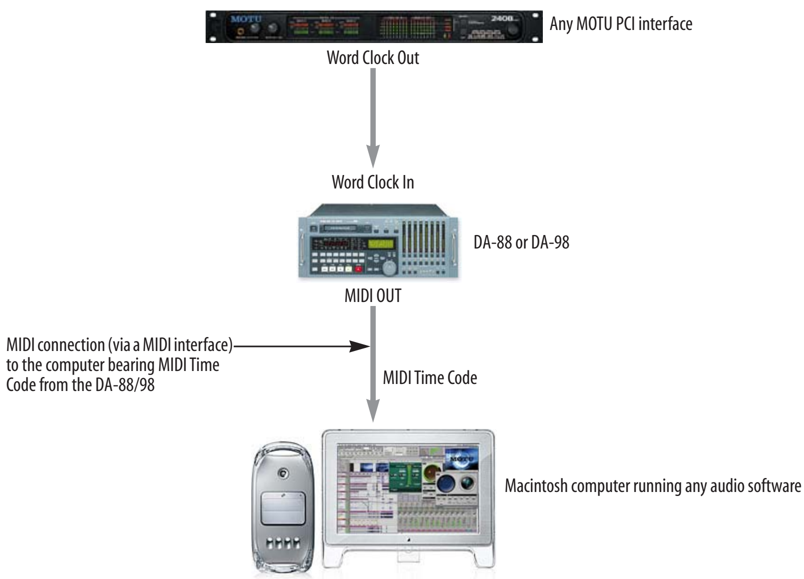
This setup provides:

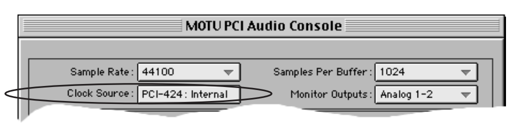
✗ No sample-accurate locating.

✗ No transport control from the computer (transport control is from the DA-88/98 itself).

✓ Timing accuracy that is at least as good as MIDI time code (quarter-frame)— and maybe even tighter.

✓ Continuous sync to SMPTE time code on the time code track of the tape in your DA-88/98.





With this setup, in the MOTU PCI Audio Console window, you can choose any clock source, since the DA-88/98 is slaving to the 2408mk3. The DA-88/98 will slave to the 2408mk3, regardless of what the PCI-424 system is slaving to.

Figure 3-27: This setup shows how you can slave your audio software to the time code track on a Tascam DA-88/98 tape. This is the simplest case, with no other devices involved. But regardless of your setup, make sure that the DA-88 and PCI-424 system share the same audio clock.

SYNCING ADAT 'LIGHTPIPE' DEVICES

The word *lightpipe* is our short-hand way of referring to any device that connects to a 2408mk3 interface via an ADAT optical cable. But we make a further distinction: a lightpipe device is also one that doesn't care about sample location. An example is a digital mixer. Since the digital mixer is not a recording device, it has no sense of sample location like an ADAT. An ADAT can cue to a specific sample number (e.g. sample number 43,478, 103) — as can any device that supports ADAT sync, but a digital mixer simply mixes and processes audio digitally, with no sense of location in time. There are many other devices that fall into this category, including digital effects processors, synthesizers, A/D converters, and many more.

For Alesis recorders or other devices that support ADAT sync, sync them with a 2408mk3 as described in the previous sections of this chapter.

For *lightpipe* devices, such as digital mixers, all you have to do is make sure that their digital audio clock is phase-locked (in sync with) the 2408mk3. There are three ways to do this:

- Slave the lightpipe device to the 2408mk3
- Slave the 2408mk3 to the lightpipe device
- Slave both the lightpipe device and the 2408mk3 to a third master clock (such as a Digital Timepiece or MIDI Timepiece AV synchronizer)

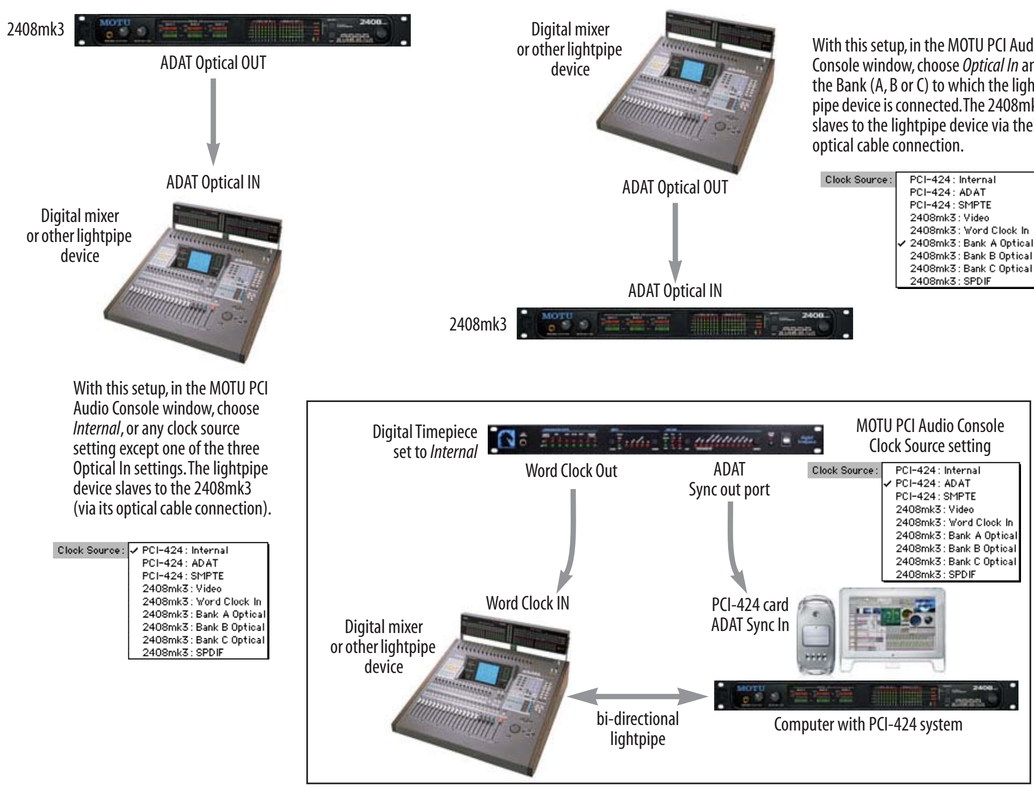


Figure 3-28: Three setups for synchronizing a lightpipe device with a 2408mk3. You can slave the lightpipe device to the 2408mk3 or vice versa with their optical connections. For more elaborate setups, you can slave both to a digital audio synchronizer like the Digital Timepiece. Don't use any of these setups for an ADAT or other optical device that records. Instead, see "Sample-accurate ADAT sync" on page 38.

SYNCING TASCAM 'TDIF' DEVICES

The acronym *TDIF* is our short-hand way of referring to any device that connects to a 2408mk3 via a Tascam TDIF digital I/O cable. But we make a further distinction: a TDIF device is also one that doesn't care about sample location. An example is the Tascam DM-24 digital mixer. Since the DM-24 is not a recording device, it has no sense of sample location like a Tascam recorder does. A Tascam recorder can cue to a specific sample number (e.g. sample number 43,478, 103) — as can any device that supports Tascam sync, but the DM-24 digital mixer simply mixes digital audio, with no sense of location in time.

For Tascam recorders, such as the MX-2424, or other devices that support Tascam Sync, connect them as described in the previous sections of this chapter.

For TDIF devices, all you have to do is make sure that their digital audio clock is phase-locked (in sync with) the 2408mk3. However, unlike the ADAT optical format, the TDIF format does not carry any form of word clock along with the actual audio signal. Therefore, sync cannot be achieved by the TDIF connection. It is always necessary to synchronize TDIF devices with the 2408mk3 via one of the following sync connections:

- Word clock
- Tascam sync (if the TDIF device supports it)

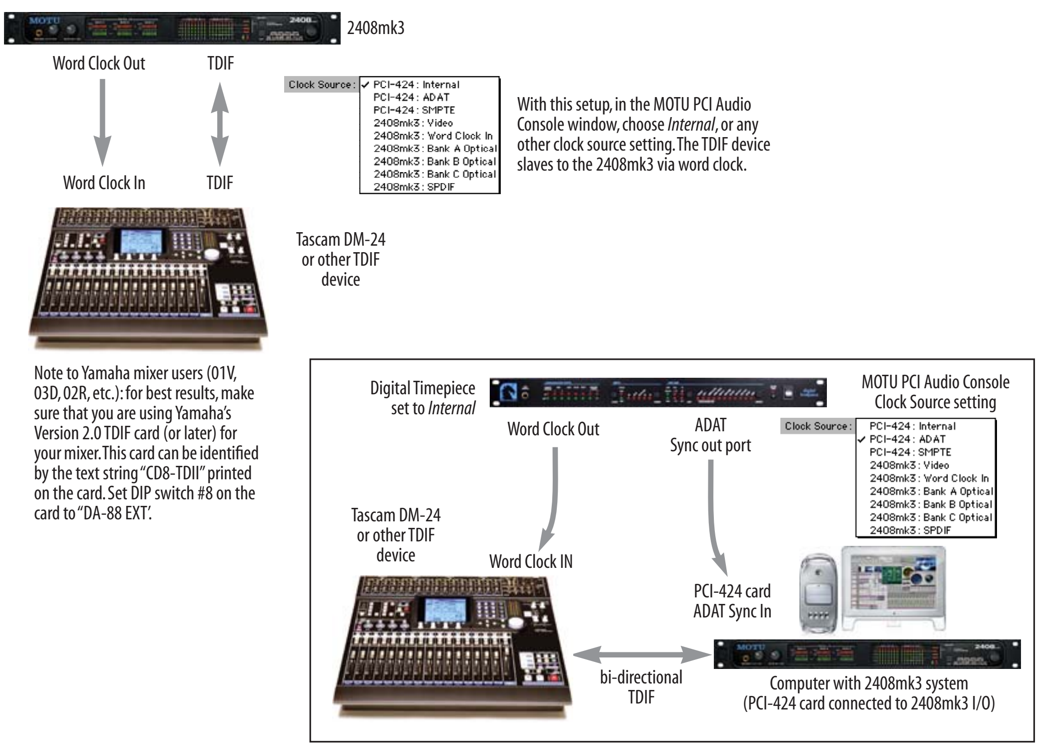


Figure 3-29: Two setups for synchronizing a TDIF device with a 2408mk3. You can slave the TDIF device to word clock from the 2408mk3 or vice versa (not shown). For more elaborate setups, you can slave both to a digital audio synchronizer like the Digital Timepiece. Don't use any of these setups for a DA-38/88/98 or other TDIF device that supports Tascam sync. Instead, see the Tascam-related sections earlier in this chapter.

SYNCING SPDIF DEVICES

DAT decks and other SPDIF devices will sync to a 2408mk3 interface in one of two ways:

- Via the SPDIF connection itself
- Via word clock

SPDIF devices with no word clock

If your DAT deck or other SPDIF device has no word clock sync connectors, just connect it to the 2408mk3 via the SPDIF connectors. When the SPDIF device is programmed to receive digital audio on its input (from the 2408mk3), it will simply synchronize to the clock provided by the audio input.

On the other hand, when you transfer audio from the SPDIF device into the 2408mk3, you'll have to slave the 2408mk3 to the SPDIF input. If you have other digital audio devices connected to the 2408mk3, and they are not slaved directly to the 2408mk3 itself, you may hear clicks and pops resulting from their unsynchronized audio clock. If so, just turn them off during the transfer.

SPDIF devices with word clock

If your SPDIF device has a Word Clock input, connect the 2408mk3's word clock Output to the SPDIF device's Word Clock input. You can then freely transfer audio between the 2408mk3 and the SPDIF device, regardless of how the rest of your 2408mk3 rig is synchronized.

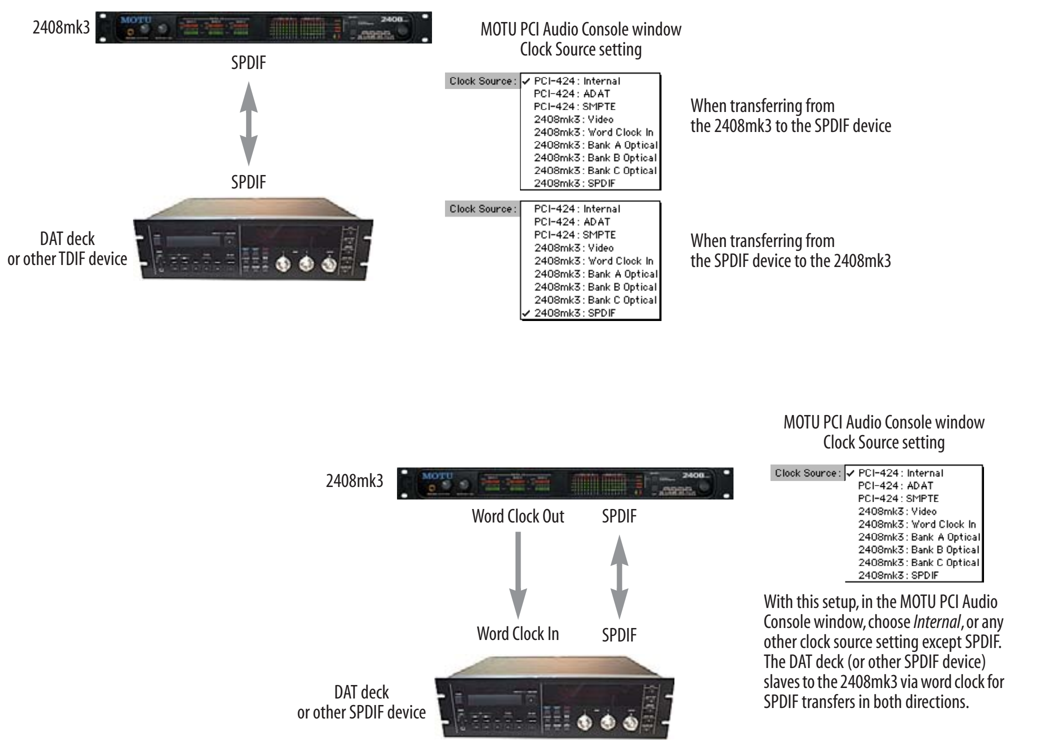


Figure 3-30: Two setups for synchronizing a SPDIF device with a 2408mk3. In the top diagram, sync is achieved via the SPDIF connection itself. In this case, you have to choose SPDIF as the PCI-424 system's clock source when recording from the SPDIF device. If you don't want to have to worry about switching the Clock Source setting depending on the direction of the SPDIF transfer, you can slave the SPDIF device to word clock from the 2408mk3 or vice versa (not shown). The Word Clock connection maintains sync, regardless of the direction of the transfer.

SYNCING WORD CLOCK DEVICES

All MOTU PCI audio interfaces provide a word clock input and/or output that allows you to synchronize the PCI-424 system clock with other word clock devices.

The HD192 interface has two word clock inputs: *System Word* and *AES Word In*. Use the *System Word Input* for general purpose word clock sync. For details about the *AES Word In*, see “Clocking scenarios for AES/EBU input” on page 32.

the 24I/O interface provides one BNC *Word Clock* jack that can be switched via software between input or output. For details, see “24I/O interface options” on page 72.

For standard word clock sync, you need to choose an audio clock master (as explained in “Choosing a digital audio clock master” on page 35). In the simplest case, you have two devices and one is the word clock master and the other is the slave as shown below in Figure 3-31 and Figure 3-32.

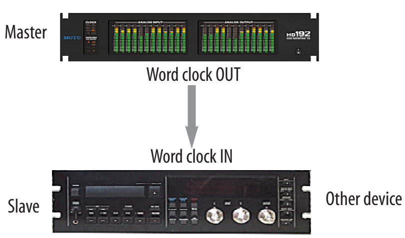


Figure 3-31: Slaving another digital audio device to the PCI-424 system via the word clock of the PCI interface (an HD192 in this example). For the PCI-424 system clock source, choose any source besides word clock, as it is not advisable to chain word clock.

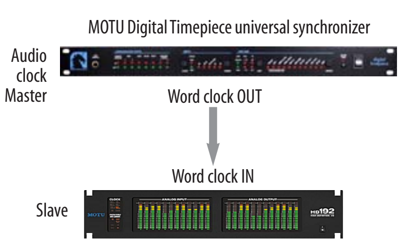


Figure 3-32: Slaving the PCI-424 system to word clock. For the PCI-424 system clock source, choose the word clock input of the interface receiving the word clock.

Don’t chain word clock

If you have three or more digital audio devices that you need to synchronize, avoid chaining their word clock connections (OUT to IN, OUT to IN, etc.), as this causes problems. Instead, use a dedicated synchronizer like the Digital Timepiece or a word clock distribution device of some kind.

Slaving to 2x and 1/2x word clock

All MOTU PCI interfaces that support 96 kHz operation (see “Sample rate” on page 66 for a summary) have the ability to slave to a word clock signal running at either twice or half the current PCI-424 system clock. For example, the PCI-424 system could be running at 96 kHz while slaving to a 48 kHz word clock signal from a MOTU Digital Timepiece. Similarly, the PCI-424 system could run at 88.2 kHz and slave to 44.1 kHz word clock. Conversely, the PCI-424 system could run at 48 kHz and slave to a 96 kHz word clock signal. In all of these cases, the front panel clock LEDs flash both sample rates to indicate that the PCI-424 system is slaving to word clock at either twice or half its system clock rate. But if the PCI-424 system is running at 96 kHz, it cannot slave to word clock running at 44.1 kHz.

Remember, the word clock signal must be one of the following:

- the same as the PCI-424 system clock
- twice the PCI-424 system clock
- half of the PCI-424 system clock

SYNC FOR 2408MK3 STAND-ALONE OPERATION

The 2408mk3 interface goes into *stand-alone* mode whenever it is not under control of the PCI-424 software driver. In other words, whenever there isn't any 2408mk3-related software running on the computer, the 2408mk3 I/O will operate as a stand-alone format converter.

For further details about this, see chapter 5, "2408mk3 Front Panel Operation" (page 57).

In most cases, when the 2408mk3 is performing stand-alone format conversion, the source of the transfer should serve as the clock source. Many of the synchronization scenarios already discussed in this chapter can be used. Below is a summary. For details on how to make these settings from the front panel, see "Clock" on page 60 and "Source" on page 62.

Front panel SOURCE setting	Front panel CLOCK setting	Required sync connections
ADAT Bank A	Dig (Digital)	ADAT optical (OUT from the source device to IN on the 2408mk3)
ADAT Bank B	Dig (Digital)	same as above
ADAT Bank C	Dig (Digital)	same as above
ADAT Bank A, B, C	Dig (Digital)	same as above
Tascam A	Wrd (word clock) (with no synchronizer) OR PCI (Use *PCI* if the 2408mk3 is slaved to a synchronizer via ADAT sync on the PCI-424 card, and the Tascam source is also slaved to the synchronizer) Note: Tascam itself cannot be used as the clock source, since TDIF carries no clock	Figure 3-29 on page 47
Tascam B	Same as above	Figure 3-29 on page 47
Tascam C	Same as above	Figure 3-29 on page 47
Tascam A, B, C	Same as above	Figure 3-29 on page 47
SPDIF	Dig (Digital) OR Wrd (Word clock)	Figure 3-30 on page 48
Analog	Any setting that synchronizes the 2408mk3 with the destination format	ADAT: Figure 3-28 on page 46 Tascam: Figure 3-29 on page 47 SPDIF: Figure 3-30 on page 48

SYNCING LARGE SYSTEMS

If you are connecting the PCI-424 to a lot of other digital audio gear, get a Digital Timepiece. It can synchronize a wide variety of devices, and it offers sample accurate synchronization for devices that support it. You will also be able to control everything from the transport controls of your audio software. If you have even more devices than a single Digital Timepiece can support, consider a word clock distribution device. Products like this offer multiple word clock outputs.

CHAPTER 4 Installing the PCI-424 Macintosh Software

OVERVIEW

The PCI-424 system ships with this software:

Software component	Location	Purpose
MOTU Folder	Extensions Folder	Contains the drivers and hard disk recording engine for the PCI-424 card and hardware. Required for 2408mk3 operation.
MOTU PCI Audio Console	Top level of your startup disk	Provides access to all of the settings in the PCI-424 card and the audio interfaces connected to it. Required for PCI-424 operation.
MOTU SMPTE Console	Top level of your startup disk	Provides access to the PCI-424 system's SMPTE time code & video sync features.
AudioDesk Workstation Software	Top level of your startup disk	Provides complete multi-track recording, mixing and processing. Optional.
ASIO MOTU PCI Driver	In the ASIO Drivers folder of Cubase or any other ASIO compliant audio program	Allows Cubase or other ASIO-compliant software to do multi-channel input and output with the 2408mk3. Only required if you are using Cubase or another ASIO-dependent program.
MOTU PCI Sound Manager Driver	Extensions Folder	Allows any Sound Manager compatible software to do stereo input and output with the 2408mk3 hardware. Optional.
AudioDesk Demo Project	Anywhere you want	Provides a multi-track mix that you can open, play, and mix in AudioDesk. Optional.
MOTU Setup Wizard	Top level of your startup disk	Asks you questions about the gear you'll connect to the 2408mk3, and then helps you set everything up, including the software
CueMix Console	Top level of your startup disk	Gives you complete control over the PCI-424's CueMix DSP feature, which provides no-latency monitoring of live inputs through your PCI-424 system.

RUNNING THE MOTU AUDIO/AUDIODESK SOFTWARE INSTALLER

Insert the Macintosh and Windows compatible installer CD-ROM into your Macintosh and run the MOTU Audio/AudioDesk installer on the CD. It will guide you through the installation. If you are unsure about what components to install, refer to the rest of the sections in this chapter, which explain the purpose of each software item.

SETUP WIZARD

Check out the Setup Wizard. It helps you figure out how to connect your gear to the PCI-424 audio interfaces, and it even configures the PCI-424 driver for you based on your setup.

MOTU FOLDER

The MOTU Folder goes in the Extensions Folder inside the System Folder on the Macintosh startup disk.

The MOTU Folder contains the "under the hood" software drivers and hard disk recording engine for the PCI-424 hardware. It also holds the real-time effects processing plug-ins used by MOTU Audio system compatible software, such as AudioDesk and Digital Performer.

The items in the MOTU Folder are required for PCI-424 operation.

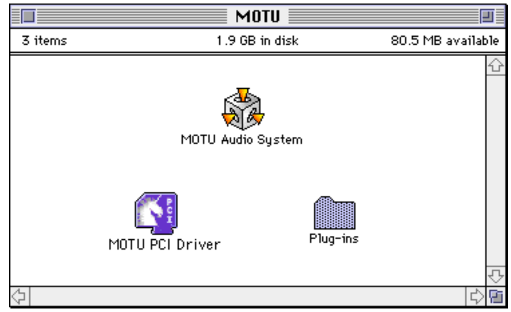


Figure 4-1: The MOTU Folder contains the software driver for the PCI-424 card and the 'MOTU Audio System' hard disk recording engine.

MOTU PCI AUDIO CONSOLE

The MOTU PCI Audio Console application is placed by the PCI-424 software installer on the top level of your Mac's startup disk. It gives you access to all of the basic settings in your PCI-424 card and the interfaces connected to it, such as clock source settings and routing to and from the computer.

For complete details, see chapter 6, "MOTU PCI Audio Console" (page 65).

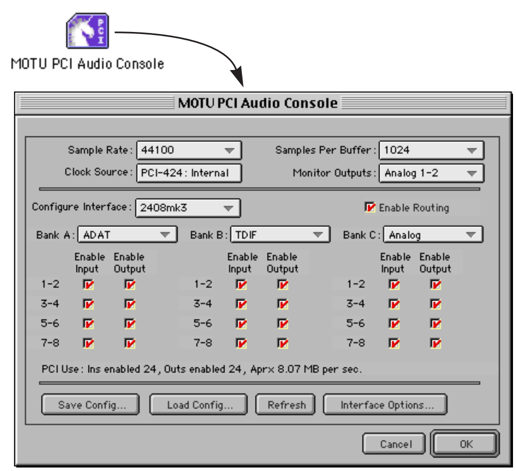


Figure 4-2: The MOTU PCI Audio Console application gives you access to all of the settings in the PCI-424 card and the audio interfaces connected to it.

AUDIODESK WORKSTATION SOFTWARE

The PCI-424 installer places AudioDesk on the top level of your Macintosh's startup volume.

AudioDesk is an advanced workstation software package for the PCI-424 system that lets you record, edit, mix, process, bounce and master multi-track digital audio recording projects. Advanced features include real-time 32-bit effects processing, sample-accurate synchronization with ADATs and Tascam tape decks, 24-bit recording, and much more.

See the AudioDesk manual included with your system for details.

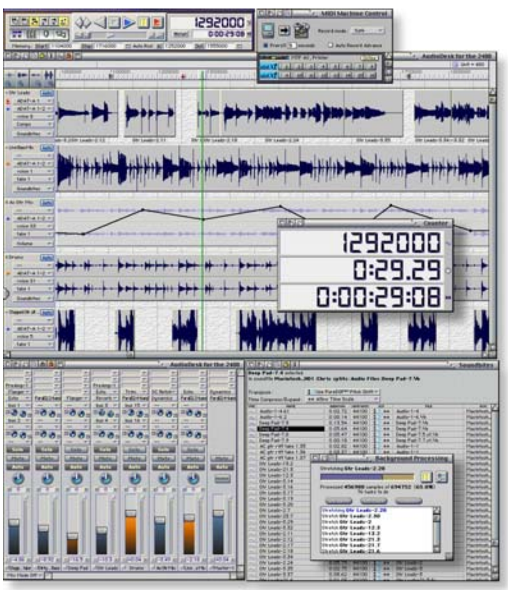


Figure 4-3: AudioDesk.

ASIO MOTU PCI DRIVER

ASIO stands for *Audio Streaming Input* and *Output*. The ASIO MOTU PCI driver allows the PCI-424 audio card to do multi-channel input and output with Steinberg's Cubase software, or any other audio application that supports ASIO drivers.

The ASIO MOTU PCI driver is only required if you are using Cubase or other audio software that depends on the ASIO driver to do multi-channel I/O with the PCI-424 system).

Digital Performer and AudioDesk do not use the ASIO MOTU PCI driver.

The ASIO MOTU PCI driver should be placed in the ASIO folder of Cubase or other ASIO-compliant software.

For details about using Cubase with the PCI-424 system, see chapter 9, "Cubase, Nuendo and other ASIO Software" (page 89).

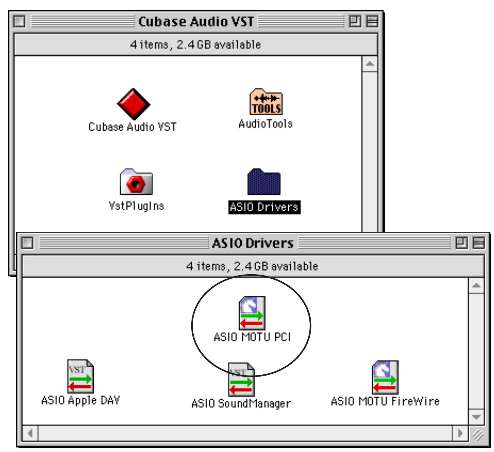


Figure 4-4: The ASIO MOTU PCI Driver.

MOTU PCI SOUND MANAGER DRIVER

The MOTU PCI Sound Manager driver is placed by the software installer in the Extensions Folder of the System Folder on your Mac's startup disk. This system extension allows the Apple Sound Manager to use the PCI-424 hardware for stereo input and output. In turn, any Sound Manager compatible software can also use the PCI-424 hardware for stereo input and output.

☛ Sound Manager only supports stereo input and output. Therefore, Sound Manager applications can only record and play through two inputs and outputs at a time.

For details about Sound Manager, see chapter 10, "Sound Manager" (page 97).

Figure 4-5: The MOTU PCI Sound Manager Driver.

CUEMIX CONSOLE

This program provides a mixing console that gives you control over the PCI-424 card's no-latency CueMix DSP features. For details, see "CueMix DSP hardware monitoring" on page 108.

CHAPTER 5 2408mk3 Front Panel Operation

OVERVIEW

The 2408mk3 interface has two modes of operation:

- PCI mode
- Stand-alone mode

In PCI mode, the 2408mk3 is under complete control of the PCI-424 card and any 424-related software running on your computer. The front panel LEDs serve entirely as status indicators.

In Stand-alone mode, the 2408mk3 front panel buttons and LEDs give you control over the 2408mk3's stand-alone format conversion features.

PCI MODE

STAND-ALONE OPERATION

SYNC

METERING

PCI MODE

The 2408mk3 goes into PCI mode as soon as you launch any program that uses the PCI-424 software driver. (The 2408mk3 must, of course, be connected to the PCI-424 card with an Audio Wire cable.) Here are some examples of how you can put the 2408mk3 into PCI mode:

- Launch the MOTU PCI Console
- Launch AudioDesk
- Launch Digital Performer
- Launch any other host audio program that uses a PCI-424 driver
- Choose a PCI-424 driver from within your host audio program
- Start up your Macintosh after installing the MOTU PCI Sound Manager driver in your Extensions folder, which gets loaded as the machine boots up

What is PCI mode?

In PCI mode, the 2408mk3 I/O is completely under control of the PCI-424 card, and you can only make changes to its settings with the MOTU PCI Audio Console software. You still have access to all of its features, but in PCI mode, you control it entirely from the PCI Audio Console window running on the computer, not from the front panel buttons.

Analog metering vs. clock settings

In PCI mode, the only front panel button that does anything is the SELECT button, which lets you switch the ANALOG OUT bank of LEDs between analog output metering and a display of the 2408mk3's current clock settings as shown on the next page in Figure 5-1.

Clock status

In PCI mode, you know that the 2408mk3 is displaying its clock status when all three CLOCK, SOURCE, and BOUNCE LEDs are illuminated as shown in Figure 5-1. If they aren't, press the *SELECT* button. In this mode the top two rows of LEDs (*CLOCK*) tell you the sync settings in the 2408mk3 hardware as determined by the MOTU PCI Audio Console software running on the computer. To change the settings, use the PCI Audio Console window as explained in "Clock Source" on page 66.

When the sample rate LEDs flash

In PCI mode, the 48/44.1 sample rate LEDs flash when the 2408mk3 has no clock from the currently selected clock source (as chosen in the PCI Audio Console window). For example, if you choose a PCI-424 clock source setting in the console (*Internal* or *ADAT*), the *PCI* LED becomes illuminated. *PCI* indicates that the 2408mk3 interface is slaved to the PCI-424 card. But if you then switch off the computer, the 2408mk3 has no clock from the card, and the 48/44.1 lights flash to alert you that the 2408mk3 interface is not receiving a digital audio clock. As another example, if you've chosen *PCI-424: ADAT* as the Clock source in the PCI Audio Console window, but your ADAT is turned off, the LEDs will also flash.

Analog metering

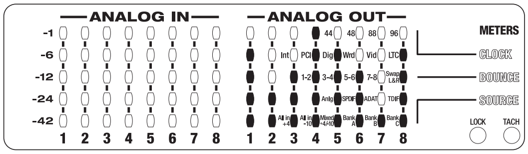
In PCI mode, press the SELECT button (if necessary) to illuminate the METERS LED as shown at the top of Figure 5-1. This LED indicates that the LEDs in the ANALOG OUT section are serving as five-segment output level meters.

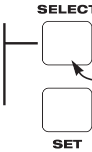
STAND-ALONE OPERATION

The 2408mk3 I/O goes into stand-alone mode whenever it is not under control of the PCI-424 software driver. In other words, whenever there isn't any 2408mk3-related software running on the computer, the 2408mk3 I/O will operate as a stand-alone format converter.

Here are some ways in which you can put the 2408mk3 into stand-alone operation:

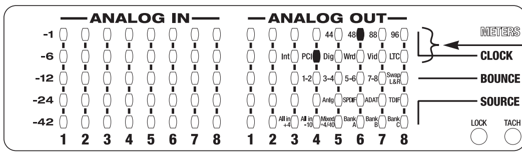
PCI Mode, Analog metering display





Use the SELECT button to toggle the ANALOG OUT bank of LEDs between the analog metering display (above) and the clock status display (below). If the 2408mk3 has no clock, it won't switch out of the clock display (into metering).

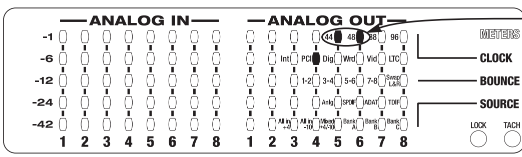
PCI Mode, Clock display



In PCI mode, these two rows display the current clock settings in the 2408mk3 (as set in the PCI Audio Console software on the computer).

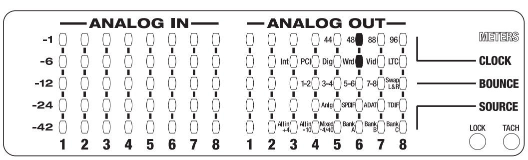
These three LEDs indicate that the 2408mk3 is in PCI mode and displaying its current clock settings (as chosen in the PCI Audio Console software).

PCI Mode, Clock display, No clock

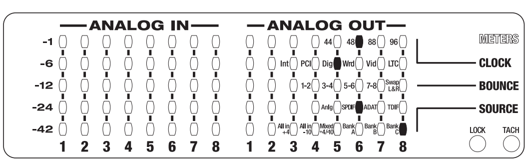


The two sample rate LEDs flash when the PCI-424 system cannot detect a digital audio clock. For example, if you've chosen ADAT sync as the Clock Source in the PCI Audio Console window, but there is no ADAT connected (or it is turned off), the 44.1 and 48 kHz LEDs will flash.

PCI Mode, Clock display, locked to word clock

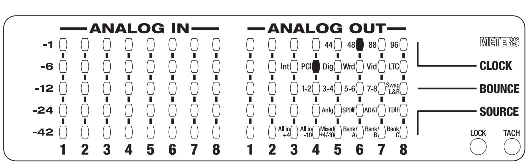


PCI Mode, Clock display, locked to SPDIF



In this example, the clock source is the SPDIF digital input of the 2408mk3 (which is active on Bank C).

PCI Mode, Clock display, locked to the PCI-424 card (Internal or ADAT)



"PCI" means that the 2408mk3 is slaved to the PCI-424 card's audio clock.

Figure 5-1: In PCI mode, the 2408mk3 settings can only be changed from the computer, and the SELECT button toggles the ANALOG OUT bank of LEDs between analog output metering and a status display of the 2408mk3's current settings.

- Shut down the computer.
- Quit AudioDesk, Digital Performer or any other software that uses a PCI-424 driver.

☛ If you have installed the MOTU PCI Sound Manager Driver Extension on your Macintosh, it will put the 2408mk3 I/O into PCI mode as soon as the computer starts up (as soon as the PCI Sound Manager Extension is loaded). So in this case, you'll need to shut down the computer entirely.

- Unplug the Audio Wire cable from the 2408mk3 (although this is not generally recommended).

Stand-alone format conversion
In stand-alone mode, the 2408mk3 I/O acts as a digital audio format converter. It lets you transfer audio from ADAT optical to Tascam TDIF, or vice versa, up to 24 channels at a time. It also lets you transfer ADAT or Tascam to SPDIF, or vice versa. Several basic track shifting options are provided.

The SELECT button
In stand-alone mode, the SELECT button lets you cycle through the four LEDs to its left, as shown below. These LEDs blink continuously to confirm that the 2408mk3 is in stand-alone mode.

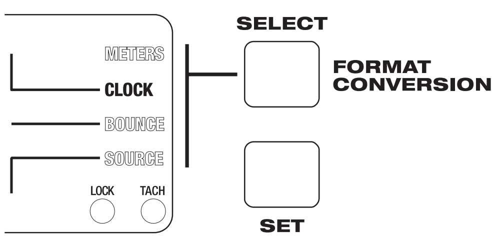
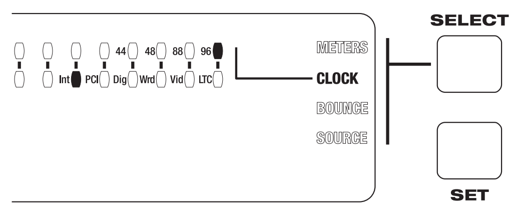


Figure 5-2: The SELECT button cycles through the four LEDs, each of which lets you make a format conversion setting with the SET button. The METERS setting switches the LEDs in the ANALOG OUT section over to output metering, as described in "Analog metering" on page 58.

No clock
In stand-alone operation, you specify a clock source as explained in the next section. However, if you choose an external clock source, and the 2408mk3 does not detect a clock signal, the 44.1 and 48 kHz sample rate lights flash to alert you that no signal can be detected. For example, if you've chosen SPDIF as the clock source (as explained in the next section), but there is no SPDIF signal detected, the 44.1 and 48 kHz LEDs will flash.

Clock
The CLOCK setting determines the digital audio clock source for the 2408mk3 when it is operating as a stand-alone format converter. Press the SELECT button repeatedly until the CLOCK LED is illuminated. Then press the SET button repeatedly to make your clock setting choice. Notice that for the *Internal* setting, you also specify the sample rate (44.1, 48, 88.2 or 96 kHz). For the *Dig* (Digital) setting, you choose between low and high sample rates (44.1/48 and 88.2/96, respectively). The same is true for the *Wrd* (word clock) setting.

For further information about what clock settings to choose, see "Sync for 2408mk3 stand-alone operation" on page 50.



Sample rates (44.1, 48, 88.2, 96)
The top row of LEDs let you specify the sample rate for a stand-alone transfer for the following clock sources: Internal (*Int*), Digital (*Dig*) and Word Clock (*Wrd*). Just press the SET button repeatedly to cycle through the desired sample rates.

For the *Int* setting, the SET button cycles through all four possible sample rates, letting you choose the desired rate.

For the *Dig* and *Wrd* settings, however, the SET button cycles through low and high sample rate settings only (44.1/48 and 88.2/96, respectively). Choose the pair that corresponds to the sample rate of the incoming clock source.

☛ At high sample rates (88.2 and 96 kHz), the 2408mk3 does not support stand-alone transfers to or from the SPDIF input or output. Use CueMix Console for digital transfers to and from SPDIF.

Int (Internal)
Use the 2408mk3's own *Internal* clock when you are using Analog as the source of the transfer. Press the SET button repeatedly to choose the desired sample rate.

PCI
The *PCI* setting does not apply to stand-alone operation, so it is not available as a clock source in stand-alone mode.

Dig (Digital)
The *Dig* (Digital) setting refers to the digital device you have chosen in the SOURCE row of LEDs as the source of a digital transfer. For example, it might be an ADAT optical device connected to Bank A or a SPDIF device on Bank C. If you are not using any other means to synchronize the external devices you are transferring between, use the *Dig* setting, which makes the 2408mk3 slave to the digital input source. (The destination device can slave to the output from the 2408mk3.)

Use the *Dig* setting if you are using SPDIF as the source of the transfer.

The *Dig* setting has two possible sample rates: low (44.1/48) or high (88.2/96). Press the SET button repeatedly to choose the setting that matches the sample rate of the digital input clock source.

Wrd (word clock)
Use the *Wrd* clock source setting when you are slaving the 2408mk3 to its word clock input. For example, if you have the 2408mk3 synchronized to a word clock device that is also serving as a word clock master for the digital audio device you have chosen as the source of the digital transfer.

☛ Only use the *Wrd* setting if the 2408mk3 is slaved to the same word clock source as the device that is the source of the transfer, either directly from the device itself or from a third device that is serving as a word clock master.

The *Wrd* setting has two possible sample rates: low (44.1/48) or high (88.2/96). Press the SET button repeatedly to choose the setting that matches the sample rate of the word clock source.

Vid (Video)
The *Vid* setting does not apply to stand-alone operation, so it is not available as a clock source in stand-alone mode.

LTC (Longitudinal Time Code)
The *LTC* setting does not apply to stand-alone operation, so it is not available as a clock source in stand-alone mode.

Bounce
The BOUNCE setting lets you shift all tracks during the transfer. Press the SELECT button repeatedly until the BOUNCE LED is illuminated. Then press the SET button repeatedly to make your bounce setting.

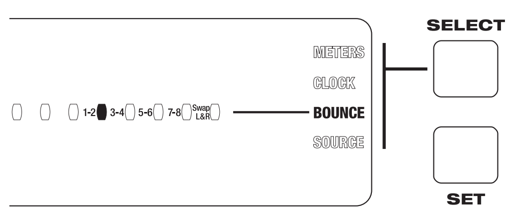


Figure 5-3: The Bounce setting lets you shift tracks during a transfer.

In most cases, when you make a digital audio transfer, you won't need to bounce, as you'll want the destination tracks to match their source tracks.

There may be times, however, when you'd like to shift tracks. For example, lets say that you had a stereo guitar part on ADAT tracks 3-4, and you'd like to transfer them to Tascam tracks 1-2. To accomplish this, you would use the Bounce setting shown below.

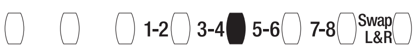


Figure 5-4: Shifting tracks 3-4 to tracks 1-2.

All other tracks are shifted in a similar fashion.

For example, if the 2408mk3 is mapping channels 3-4 to 1-2, channels 1-2 are wrapped back around to the top of the bank to channels 7-8. the rest of the channels are shifted in a similar fashion.

Swap L to R (Left to Right)
For each track shift option, you have the choice of making a straight shift, such as that shown in Figure 5-4 above, but if you press the Set button one more time, you are also given the choice of swapping the left and right channels of each track pair, as demonstrated below.

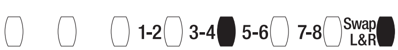


Figure 5-5: Swapping the left and right channels, in addition to shifting the tracks.

In Figure 5-5, not only is the 2408mk3 shifting tracks 3-4 to tracks 1-2, it is also swapping the channels within each pair. So channel 4 is actually being mapped to channel 1, and channel 3 is being mapped to channel 2. This gives you enough flexibility to map any channel to any other channel.

Swapping is applied to all other track pairs as well.

Source

The SOURCE setting determines the audio source for the transfer. Press the SELECT button repeatedly until the SOURCE LED is illuminated. Then press the SET button repeatedly to make your choice.

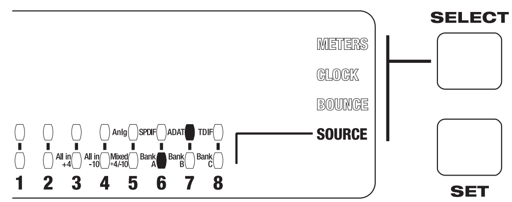


Figure 5-6: Choosing a source for a transfer. Press the SET button repeatedly to choose which audio format you want to record from, and from which bank(s) (A, B or C, or all three combined).

☛ At high sample rates (88.2 and 96 kHz), the 2408mk3 does not support stand-alone transfers to or from the SPDIF input or output. Therefore, if you choose SPDIF as the source, then there will be no output to the ADAT or TDIF banks. Conversely, if you choose ADAT or TDIF as the source, there is no output sent to SPDIF. Use CueMix Console for digital transfers to and from SPDIF.

Anlg (Analog)
When you choose Analog as the source in stand-alone mode, the analog bank is hard-wired to Bank C (this is arbitrary and doesn't matter because there is only one bank of analog). With Analog as the source, the Analog bank gets sent to (duplicated on) all three output banks (A, B and C) in both ADAT and Tascam formats. In addition, channels 1-2 of the Analog input bank are sent to the SPDIF outputs.

There are three variations for the analog source setting. Press the SET button repeatedly to choose the desired variation as explained below:

Analog source setting	Explanation
All in +4	All analog inputs are set to a +4 dB input level.
All in -10	All analog inputs are set to a -10 dB input level.
Mixed +4 /-10	Analog input levels match the settings specified in the MOTU PCI Audio Console (as explained in "Input Reference Level" on page 72).

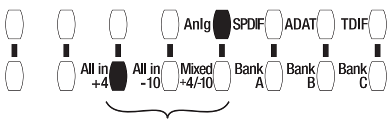


Figure 5-7: The three variations for the analog input source setting.

SPDIF

When you want to transfer from SPDIF, there is only one choice: Bank C (since SPDIF can only be used on Bank C.)

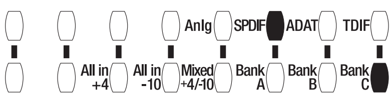


Figure 5-8: Choosing SPDIF as a source for a transfer.

The SPDIF input is mapped to the rest of the channel pairs of the bank, and then it is also mapped to all three ADAT banks and all three Tascam banks.

ADAT and TDIF

When you want to transfer from either ADAT or TDIF, you can choose Bank A, B or C, or all three banks at once, as shown below in Figure 5-9.

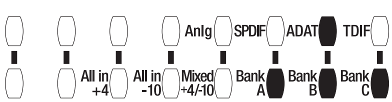


Figure 5-9: Choosing a source for a transfer. Press the SET button repeatedly to choose Bank A, B or C, or all three combined as shown.

When you choose an individual bank, it is mapped to all three output banks, duplicated on each bank. For example, if ADAT Bank B is the source, it will be sent to TDIF banks A, B and C. (It will also be sent to ADAT banks A, B and C.)

When you choose all three banks together as the source, they pass audio straight through, i.e. Bank A to A, B to B, and C to C.

Analog metering

Press the SELECT button repeatedly (if necessary) to illuminate the METERS LED as shown in Figure 5-10 below. This LED indicates that the LEDs in the ANALOG OUT section are serving as five-segment output level meters.

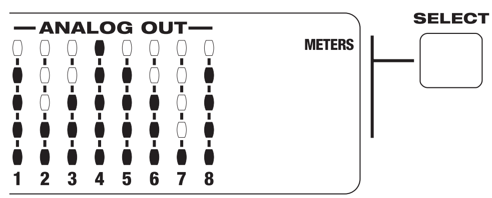


Figure 5-10: You can ignore the LED labels in the ANALOG OUT section of the front panel when the LEDs are acting as output level meters.

LOCK AND TACH

When the 2408mk3 is resolving to video or SMPTE time code (via its built-in sync features), the LOCK light glows green when lockup has been achieved. The TACH light blinks once per second when the 2408mk3 is successfully reading address (time code) information. For details, see "Syncing to SMPTE time code" on page 41 and "Syncing to video" on page 42.

DIGITAL ACTIVITY LEDs

The Activity LEDs in the ADAT/TDIF section show input and output activity on all three banks of digital input and output. A signal of -40dB or stronger illuminates these LEDs.

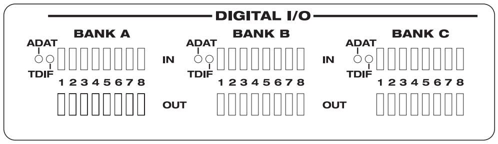


Figure 5-11: The 2408mk3's ADAT/TDIF activity LEDs are triggered by a signal of -40dB or greater.

CHAPTER 6 MOTU PCI Audio Console

OVERVIEW

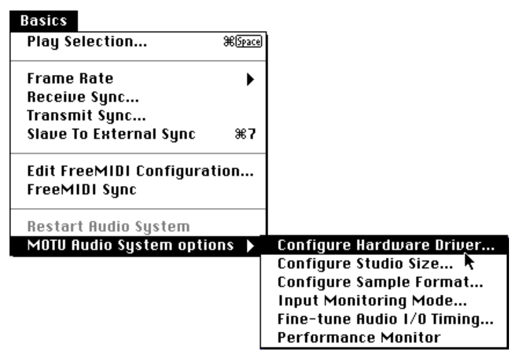
The MOTU PCI Audio Console Window gives you complete control over the settings in your PCI-424 hard disk recording system.

ACCESSING THE PCI AUDIO CONSOLE WINDOW

There are several ways to access the PCI Audio Console window:

■ From the Mac OS desktop, run MOTU PCI Audio Console (the stand-alone applet for the PCI-424 system)

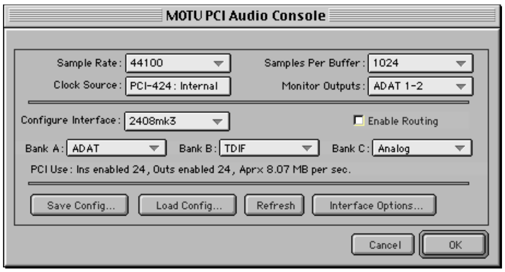
■ From within AudioDesk™ or Digital Performer™, go to the Basics menu and choose *MOTU Audio System options>Configure Hardware Driver*



It doesn't matter which way you access the window. The settings are the same either way.

QUICK REFERENCE

For a one-page overview of the MOTU PCI Audio Console, see “Quick Reference: PCI Audio Console Window” on page 15.



PCI-424 SETTINGS

The top of the MOTU PCI Audio Console window, as shown in Figure 6-1, has several settings for the PCI-424 card itself. These are settings that are not dependent on the interfaces connected to the card. Instead, they impact the function of the system as a whole.

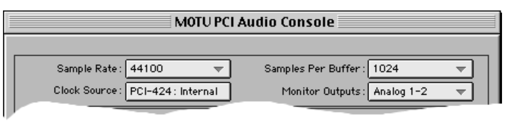


Figure 6-1: The PCI-424 settings at the top of the window are global settings that apply to the system as a whole, regardless of the MOTU PCI audio interface connected to it.

Sample rate

Choose the desired *sample rate* for recording and playback. The PCI-424 system can operate at 44.1, 48, 88.2 or 96 KHz. If an HD192 interface is connected, it can also run at 176.4 or 192 kHz. The entire PCI-424 system will run at the sample rate chosen in this menu. Interfaces that do not support the system’s current sample rate are temporarily taken off line. For example, if you run the HD192 interface(s) at 192 kHz, all other audio interfaces are taken off line. As another example, if you run a PCI-424 interface (2408mk3, 24I/O or HD192) at 96 kHz, any connected legacy PCI-324 interfaces (such as a 2408, 2408mkII, 1224 or 308) will be taken off line, with the exception of the 1296, which supports PCI-424 operation at 88.2 or 96 kHz.

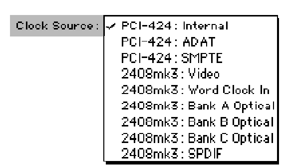
Below is a table that summarizes MOTU PCI audio interface supported sample rates:

Interface	44.1	48	88.2	96	176.4	192
HD192	✓	✓	✓	✓	✓	✓
2408mk3	✓	✓	✓	✓		
24I/O	✓	✓	✓	✓		
1296	✓	✓	✓	✓		
2408mkII	✓	✓				
2408	✓	✓				
1224	✓	✓				
24i	✓	✓				
308	✓	✓				

Make absolutely sure that all of the devices connected digitally to all PCI-424 interfaces match the PCI-424 system’s sample rate. Also make sure that your Digital Timepiece, MIDI Timepiece AV or other digital audio synchronizer matches it as well.

☛ 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 the PCI Audio Console window.

Clock Source



The *clock source* determines the digital audio clock that the PCI-424 system will use as its time base. For a complete explanation of synchronization issues, see “Make sync connections” on page 35. The following sections briefly discuss each clock source setting.

PCI-424: Internal
Use the *PCI-424: Internal* setting when you want the PCI-424 system 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. For example, you might simply be playing the hard disk tracks and mixing them to the analog outputs of your MOTU audio interface. In this case, no other digital audio clocks are involved.

Another example is transferring a mix to DAT. You can operate the PCI-424 system on its internal clock, and then slave the DAT deck to the PCI-424 system, either via the SPDIF connection (usually DAT decks slave to their SPDIF input when you choose the SPDIF input as their record source) or via the word clock output of your MOTU PCI audio interface (if your DAT deck has a word clock input).

PCI-424: ADAT
ADAT refers to the *ADAT Sync* digital audio synchronization format. It allows the entire PCI-424 system to slave to the ADAT sync chain connected to your PCI-424 audio card (via the 9-pin connector on the card). ADAT sync also carries precise, sample location information, which allows AudioDesk, Digital Performer or other sample-accurate software to transfer audio to and from ADATs without drifting by as much as one sample.

Use this setting when you are using the PCI-424 system with one or more ADATs or other ADAT Sync-compatible devices. Make sure the PCI-424 card is connected to the end of the ADAT sync chain.

You should also use this setting if you have a MOTU MIDI Timepiece AV or Digital Timepiece universal synchronizer, which allows you to drive your entire system from the transport controls of AudioDesk, Digital Performer or other computer software.

PCI-424: SMPTE
Choose this setting to resolve the PCI-424 card (and all connected interfaces) directly to SMPTE time code (LTC) being received via an analog input on any connected interface. To specify the analog input for time code, use the MOTU SMPTE Console application. For details, see “SMPTE Source” on page 119.

2408mk3: Video
Choose this setting to resolve the PCI-424 card (and all connected interfaces) directly to video being received via the Video In BNC connector on a 2408mk3 interface. (Only the 2408mk3 interface supports this feature.) In this sync mode, the system can also reference SMPTE time code via any analog input. To specify the input for time code, use the MOTU SMPTE Console application. For details, see “SMPTE Source” on page 119.

2408mk3: Word Clock In
24I/O: Word Clock In
HD192: Word Clock In
These settings refer to the Word Clock In BNC connector on their respective interface. Choosing one of these settings allows the entire PCI-424 system (including other 2408mk3 I/Os connect to the PCI-424 card) to slave to an external word clock source, such as the word clock output from a digital mixer.

The *24I/O: Word Clock In* setting is unique because it switches the operation of the single BNC word clock connector on the 24I/O interface. When you choose the *24I/O: Word Clock In* setting, the 24I/O's word clock jack becomes a word clock input. When you choose any other clock source setting, it operates as a word clock output.

2408mk3: Bank A/B/C Optical In
These settings refer to the clock provided by the ADAT optical inputs on the 2408mk3 interface. This setting can be used to slave the entire PCI-424 system directly to an optical input connection.

These settings are useful if you just need to make a simple, click-free digital transfer between the 2408mk3 and another device — where a time code reference and shared transport control are not needed — without having to set up a full-fledged synchronization scenario.

In many cases, you can set up a better operating scenario that uses one of the other synchronization options. However, there may be occasions when you have an ADAT optical compatible device that has no way of synchronizing digitally to the 2408mk3 or an external synchronizer such as the Digital Timepiece. In this case, the "Bank A/B/C Optical In" setting lets you slave the 2408mk3 to the device itself via its digital input to the 2408mk3.

Note: TDIF can't be used as a clock source
Please note that TDIF cannot be used as a clock source. This is because the TDIF I/O format does not contain a stable clock source. Therefore, if you have selected TDIF as the audio format for one of the three banks, and you then try to choose *ADAT optical* from that bank as the clock source, the PCI Audio Console will alert you that it cannot use the TDIF input as a clock source.

2408mk3: SPDIF
This setting refers to the SPDIF RCA input connector on the 2408mk3 interface. This setting allows the entire PCI-424 system (including other 2408mk3 I/Os connect to the PCI-424 card) to slave to another SPDIF device.

Use this setting whenever you are recording input from a DAT deck or other SPDIF device into the 2408mk3. It is not necessary in the opposite direction (when you are transferring from the 2408mk3 to the DAT machine).

HD192: Bank C AES/EBU
This setting refers to the AES/EBU digital input jack on the HD192 interface. This setting allows the entire PCI-424 system (including other interfaces connect to the PCI-424 card) to slave to another AES/EBU device connected to the HD192 AES/EBU digital input.

Use this setting whenever you are recording input from a DAT deck or other AES/EBU device into the HD192, and you are not using the HD192's sample rate conversion features.

HD192: Internal
24I/O: Internal
These settings produce the highest possible audio performance (best signal to noise ratio, etc.) from a PCI-424 system with one of these interfaces connected. Use this setting when you want the PCI-424 system 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. For example, you might simply be playing the hard disk tracks and mixing them to the main outputs of your audio interface. In this case, no other digital audio clocks are involved.

For all practical purposes, this clock source setting produces the same results as the *PCI-424: Internal* setting explained on page 67.

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 Console, which routes the signal directly to your speakers via hardware. For details, see chapter 13, "CueMix Console" (page 111).

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 AudioDesk, Digital Performer or other 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 Console allows you to monitor dry, unprocessed live inputs with no delay at all. For complete details, see chapter 12, "Reducing Monitoring Latency" (page 105).

Monitor Outputs

The Monitor Outputs option determines which outputs the MOTU PCI Sound Manager driver will use.

INTERFACE SETTINGS

The middle portion of the MOTU PCI Audio Console Window displays settings for the audio interface(s) connected to your PCI-424 card.

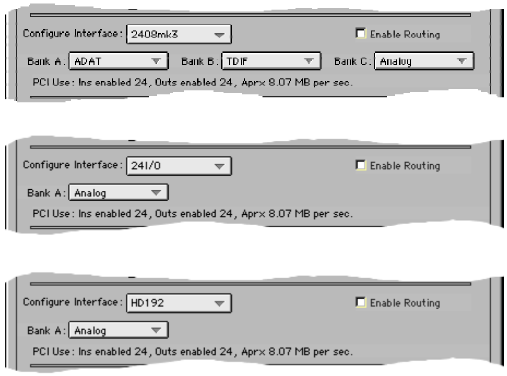


Figure 6-2: The interface settings for the 2408mk3, 24I/O and HD192 interfaces.

Working with multiple interfaces

If you have two or more audio interfaces connected to your PCI-424 card, the PCI Audio Console displays settings for one interface at a time. To view the settings for a particular interface, choose it from the *Configure interface* menu as shown in Figure 6-1 on page 66.

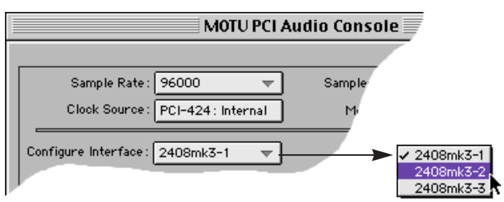


Figure 6-3: The console displays one interface at a time. Choose the desired interface from the Configure Interface menu.

Bank A / B / C (240mk3 interface)

The three 2408mk3 Bank menus (Bank A, Bank B and Bank C) let you choose the type of I/O format you would like to use for each of the 2408mk3's three internal banks of 8-channel I/O. For example, you could work with an ADAT connected to bank A, the 2408mk3's 8 analog inputs and outputs on Bank B, and SPDIF I/O on bank C. Or you could work with the ADAT optical format on all three banks. You can freely switch between formats at any time.

ADAT and TDIF banks at high sample rates

When using ADAT or TDIF banks on a 2408mk3 interface, and the PCI-424 system clock is set to a sample rate higher than 48 kHz, these banks change to four channels each, as these formats only support four channels at 88.2 or 96 kHz.

ADAT & TDIF always share the same output

If you have both TDIF and ADAT optical devices connected to the same bank, use the corresponding menu to switch between formats. Your choice, however, only affects input for the bank. For output, the 2408mk3 always sends the same material to both digital formats on a bank. For example, if you chose ADAT for Bank B, you would be able to record tracks in your computer software from the ADAT optical inputs connected to Bank B on your 2408mk3 interface. At the same time, however, any tracks you play back on Bank B would be output to both the ADAT and TDIF devices connected to Bank B.

Disabling banks to conserve resources

If you are not using a bank, it is a good idea to choose *Disable* from the bank menu to disable the bank entirely and free up PCI bus bandwidth and other system resources.

☛ The *Disable* menu item does not appear in the menu when the *Enable Routing* box is checked.

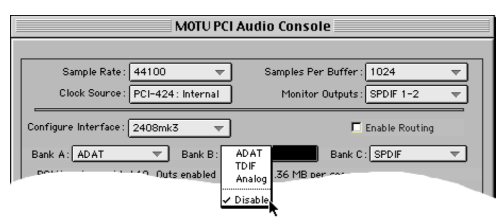


Figure 6-4: Disabling a bank to conserve system resources.

Disabling banks does not affect CueMix DSP

When you disable a bank as shown in Figure 6-4 above, it only affects PCI bus routing to the computer and your host software. Disabling a bank here has no effect on CueMix DSP mixing. Therefore, feel free to disable banks that you don't need for your software, even if you are using them for CueMix DSP live mixing and monitoring.

ENABLE ROUTING

The Enable Routing check box expands the window to display each individual input and output pair. Each check box represents the availability of that input or output pair to audio software running on your computer. Check the *Enable Input* check box to make it available as inputs to your audio software; check the *Enable Output* check box to make the output pair available to your host software. When a box is checked, the input or output pair will appear in the input or output source menus in AudioDesk, Digital Performer or other host audio application. When it is unchecked, it won't. But as described above, unchecking inputs has no effect on CueMix DSP mixing and monitoring.

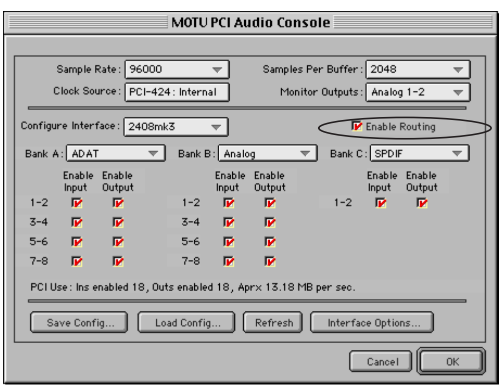


Figure 6-5: Check the 'Enable Routing' check box (circled above) to expand the window. You can then disable (or enable) individual input and output pairs.

☛ Remember: ADAT and TDIF banks show four channels (instead of eight) at sample rates higher than 48 kHz.

PCI USE

Enabled banks (or individual inputs and outputs) take up a small portion of your computer's processing power. And while the amount is quite small for individual I/O pairs, a core PCI-424 system lets you enable up to 24 inputs and outputs at a time (a total of 48). An expanded PCI-424 system with four interfaces lets you enable up to 96 inputs and outputs at a time (a total of 192). To conserve your computer's processing power, only enable banks (or individual inputs and outputs) when you are actually using them. Otherwise, leave them unchecked.

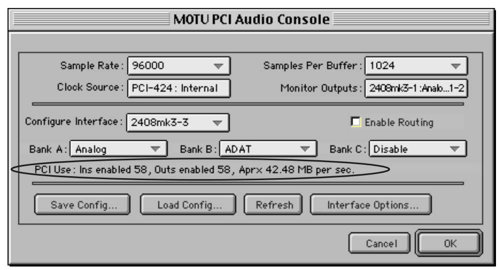


Figure 6-6: Keep an eye on the PCI bus resources when using two or more audio interfaces.

The *PCI Use* status display tells you the approximate PCI bandwidth being used by the PCI-424 card for the currently enabled inputs and outputs. This display is mostly intended for expanded PCI-424 systems, which push the limits of the PCI bus itself. A higher number will alert you that there are a significant number of inputs and outputs enabled, perhaps including the ones you are not currently viewing.

While the theoretical maximum PCI bandwidth is 132 MB per second, there are no hard and fast rules for how much bandwidth is actually available on any given computer. Many factors come in to play, including the efficiency of the bridge chip that controls the PCI bus and the number of other PCI devices on the bus competing for bandwidth. Practically speaking, most of today's personal computers seem to have approximately 30-50 MB per second of PCI bandwidth for the 2408mk3 and other PCI cards in the computer.

INTERFACE OPTIONS

The *Interface options* button opens a dialog that provides several settings for the interface currently chosen in the *Configure Interface* menu.

2408mk3 interface options

Each 2408mk3 interface has the following options:

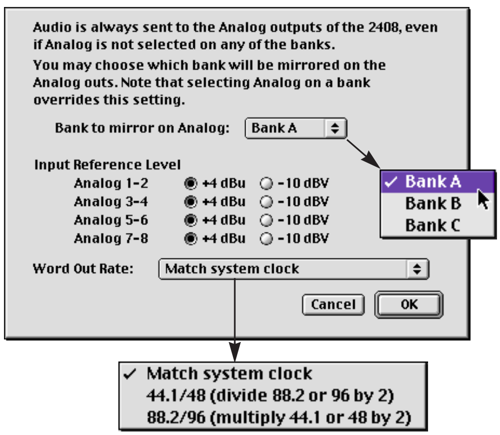


Figure 6-7: The 2408mk3 interface options.

Analog output bank mirroring

The *Bank to mirror on Analog* menu lets you choose what comes out of the Analog outputs when none of the 2408mk3's three banks (A, B or C) are set to *Analog*.

Even when *Analog* is not currently selected on any of the three banks in the PCI Audio Console window, the Analog outputs of the 2408mk3 mirror — or duplicate — one of the three banks. By default, they mirror Bank A. To choose another bank, click the *Interface Options* button and choose the desired bank from the menu as shown above in Figure 6-7.

Input Reference Level

These settings let you specific the input reference level for the four analog input pairs on the 2408mk3 interface. The factory default setting is +4 dB. If an input signal sounds weak, try the -10 dB setting.

Word Out Rate

The *Word Out Rate* menu lets you either double or halve the current system word clock rate.

If the current PCI-424 system word rate is a high sample rate (88.2 or 96 kHz), then the word clock output can either match it or operate at half the rate (44.1 or 48 kHz).

If the current system word rate is a low sample rate (44.1 or 48 kHz), the word clock output can either match it or operate at twice the rate (88.2 or 96 kHz).

24I/O interface options

Each 24I/O interface has the options shown below in Figure 6-8.

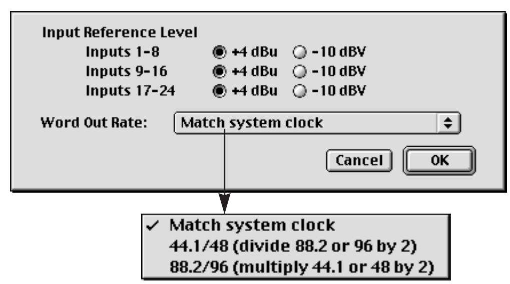


Figure 6-8: The 24I/O interface options.

Input Reference Level

These settings let you specific the input reference level for the inputs on the 24I/O interface. On the 24I/O, the input reference level is set in banks of 8 inputs at a time. The factory default setting is +4 dB. If an input signal sounds weak, try the -10 dB setting.

Word Out Rate

The *Word Out Rate* menu lets you either double or halve the current PCI-424 system word clock rate.

If the current system word rate is a high sample rate (88.2 or 96 kHz), then the word clock output can either match it or operate at half the rate (44.1 or 48 kHz).

If the current system word rate is a low sample rate (44.1 or 48 kHz), the word clock output can either match it or operate at twice the rate (88.2 or 96 kHz).

HD192 interface options

The *Interface options* button opens a dialog that provides several options for the HD192's AES/EBU section and front-panel the level meters.

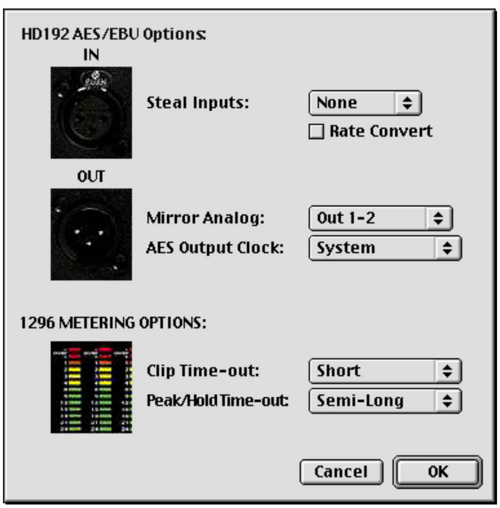


Figure 6-9: The HD192 interface options.

The HD192 AES/EBU section has flexible clocking options and separate sample rate converters on both input and output. Below is a brief summary of the possible settings for these features.

Steal inputs

Internally, the AES/EBU stereo section shares the same bus as the analog section of the HD192. The *Steal Inputs* menu lets you choose which pair of analog inputs to replace with AES/EBU input. For example, if you choose inputs 11-12 from this menu, analog inputs 11 and 12 are temporarily disabled and replaced by the AES/EBU inputs. In your host audio software, inputs 11-12 now refer to the AES/EBU input.

Rate Convert

Enable the *Rate Convert* option when you wish to convert the sample rate of the incoming AES/EBU digital audio signal to the sample rate at which the HD192 system clock is running. For example, if the HD192 system clock is running at 96 kHz, and the AES/EBU input signal is at 44.1 kHz, enable Rate Convert to transform the 44.1 kHz signal into a 96 kHz signal. Rate conversion is applied in real time as the signal passes through the AES/EBU input's sample rate converter.

Mirror Analog

Choose the analog output pair you would like to match for AES/EBU output. For example, if you choose analog Outs 3-4 from the *Mirror analog* menu, then the audio signal being sent out analog outputs 3-4 will also be sent to the AES/EBU stereo output. This would allow you, for example, to monitor a stereo mix via analog outs 3-4 while at the same time transferring the same stereo mix via AES/EBU to a DAT deck or other AES/EBU device.

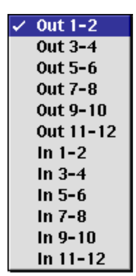


Figure 6-10: The choices in the Mirror Analog menu for the HD192's AES/EBU output.

As shown above in Figure 6-10, notice that the AES/EBU output can also directly mirror a pair of HD192 analog inputs. In effect, this lets you route any analog input pair directly to the AES/EBU output.

AES Output Clock

The AES Output Clock menu provides several clock sources and sample rate options for the AES/EBU output.

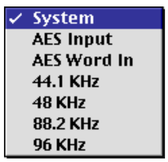


Figure 6-11: Clock sources for the AES/EBU output.

Each clock option is briefly explained below.

System

Choose *System* when you want the AES/EBU output sample rate to match the HD192's system clock, and you don't need to convert to a different sample rate or resolve the output to some other external clock. No sample rate conversion will occur when this option is chosen.

AES Input

Choose AES Input when you would like the AES/EBU output sample rate to match — and continuously resolve to — the sample rate currently being received by the AES/EBU input. If necessary, the AES/EBU output signal will be sample rate converted to match the AES input rate.

☛ Be careful when both the HD192's AES/EBU input and output are connected to the same external device: this option is likely to create a clock loop.

AES Word In

Choose *AES Word In* when you would like the AES/EBU output to match — and continuously resolve to — word clock being received from the HD192's 'AES Word In' BNC jack. If necessary, the AES/EBU output signal will be sample rate converted to match the AES Word In rate.

44.1 / 48 / 88.2 / 96 kHz

Choose one of these sample rates when you need to convert the AES/EBU output to a rate different than the HD192's sample rate. When you choose one of these rates, the additional *Fixed Frequency* option appears, as shown below in Figure 6-12.

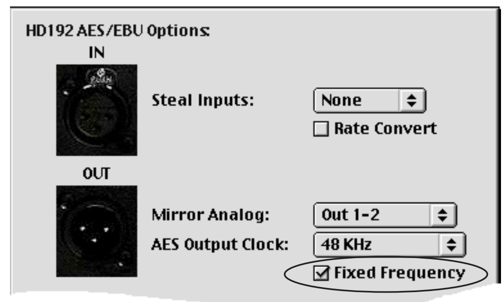


Figure 6-12: The Fixed Frequency option.

Check the Fixed Frequency option to generate the chosen sample rate with AES/EBU output independently driven by its own clock crystal (not resolved to the system clock).

Uncheck the Fixed Frequency option to generate the chosen sample rate but also resolve to the System clock. For example, if the system clock is set to 48 kHz and the AES/EBU output is set to 96 kHz, the AES/EBU output will run at exactly twice the sample rate of the system clock. This will be true even if the system clock is running faster or slower than 48 kHz because the HD192 system is slaving to an external clock.

Metering options

The HD192 provides several options for the front-panel input and output meters.

Clip Time-Out

The *Clip Time-Out* option controls how long the top red LED remains illuminated after clipping occurs (see Figure 6-13 below).

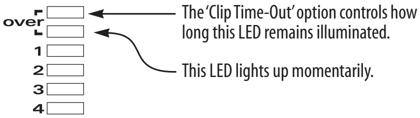


Figure 6-13: The Clip Time-Out option.

If you want the ability to clear the LED manually from your host audio software or CueMix Console, Choose *Infinite* from the *Clip Time-Out* menu. Then, in CueMix Console, click the *Clear Clip* button as shown in Figure 13-1 on page 111.

In Digital Performer or AudioDesk, you can clear the HD192 clip LEDs by choosing Audio menu>Clear All Clipping Indicators.

Peak/Hold Time-Out

The HD192 front-panel level meters support standard peak/hold metering, where the LED for the highest level recently measured on the channel remains illuminated for a brief period of time while the rest of the LEDs below it remain fully dynamic. The *Peak/Hold Time-Out* controls how long the peak-hold LED remain illuminated before going dark again.

SAVING AND RECALLING ROUTINGS

If you have several interfaces connected to your system, there are a lot of banks to configure in the middle portion of the PCI Audio Console window (one entire "page" for each). For your convenience, the *Save Config* button lets you save the current routing configuration (including the routing for the 2nd, 3rd and 4th interfaces, if you have them) so you can recall the configuration later on with an easy click of the *Load Config* button. Save the file in the standard fashion. Note that only the settings in the middle portion of the window (i.e. the input and output routings) are saved. The PCI-424 settings are not included as part of the configuration, as they are usually system-wide settings that you won't change very often.

THE REFRESH BUTTON

The *Refresh* button makes the MOTU PCI Audio Console window query the audio interfaces connected to the PCI-424 card to make sure that the settings in the window accurately reflect the settings in the hardware. Under normal operations, this should never be necessary. Even if you switch off the interfaces, the PCI-424 software driver is designed to re-establish contact with the hardware. Even so, if you suspect that the hardware might be in a different state than what you see in the window, click Refresh.

CHOOSING A CONFIG BEFORE OPENING A FILE

The settings in the PCI Audio Console window are not saved with the files you create and save with host audio applications like AudioDesk, Digital Performer, or third party audio programs. To save time, you can save the current PCI Audio Console Window settings as a file on disk (as explained earlier), along with your host audio program files. If you work on the project later on, you can quickly load the configuration file, without having to remember how the PCI Audio Console window was configured.

Some host audio applications, like AudioDesk and Digital Performer, remember the PCI Audio Console Window's input and output settings at the time you last saved a project file. This means that even if the PCI Audio Console Window doesn't currently have the exact same settings, AudioDesk and Digital Performer display the inputs and outputs that were in effect when the file was last saved. Unavailable inputs/outputs are displayed in italics.

Some third-party host applications, however, lose their input and output settings if they are not available at the time you open the file. You can save yourself the time of reassigning them by saving a 424 configuration along with the file. For details, see "Saving and recalling routings" on page 75.

CHAPTER 7 Digital Performer

OVERVIEW

Digital Performer Version 2.42 or higher is required for the PCI-424 system. Version 2.42 supports all of the advanced features of the PCI-424 system, including sample-accurate synchronization with ADATs and the Digital Timepiece.

SETTING UP YOUR SYSTEM

In addition to FreeMIDI and the other required System elements supplied by the Digital Performer installer, the PCI-424 software installer updates the drivers in the MOTU Folder (in the Extensions Folder). As described in chapter 4, “Installing the PCI-424 Macintosh Software” (page 53), the Digital Performer and 2408mk3 software installers will properly install and update everything for you. For your convenience, here is a brief summary of what Digital Performer requires to run with the PCI-424:

■ The MOTU PCI Audio Driver must be in the MOTU folder (as shown in Figure 4-1 on page 54).

■ The MOTU Audio System item in the MOTU Folder (as shown in Figure 4-1 on page 54) must be the same version as that which gets installed by the 2408mk3 installer CD-ROM (or later).

■ FreeMIDI must be installed (from either the Digital Performer or 2408mk3 installer CD).

■ If you are using a MIDI Timepiece AV or Digital Timepiece for synchronization, be sure they are present in your FreeMIDI setup.

THE MOTU PCI AUDIO CONSOLE WINDOW

The MOTU PCI Audio Console window gives you direct access to all of the settings in your PCI-424 system hardware. To open it, choose *MOTU Audio System options>Configure Hardware Driver* from the Basics menu. If you don’t see the MOTU PCI Audio Console window as shown below in Figure 7-1, choose *MOTU PCI Audio* from the menu at the top of the Configure Hardware dialog.

The PCI Audio Console window is central to the operation of your PCI-424 hardware with Digital Performer. For complete details about the important settings in this window, see chapter 6, "MOTU PCI Audio Console" (page 65).

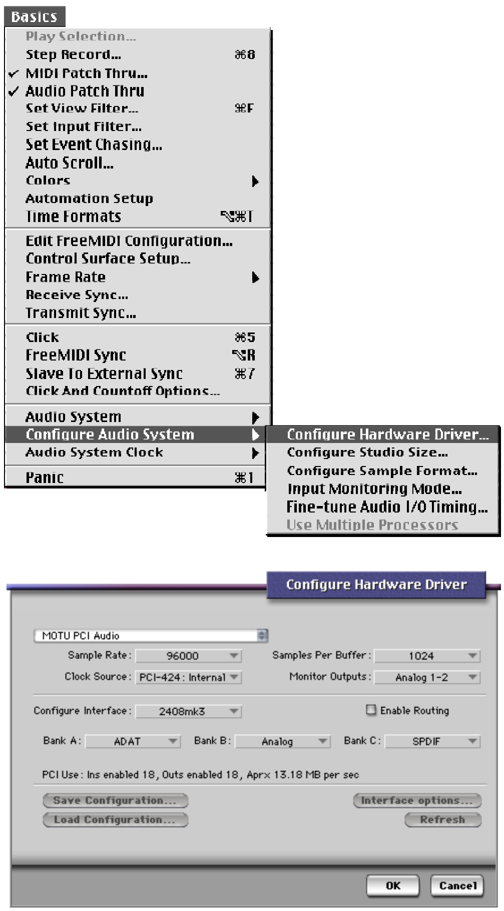


Figure 7-1: The 'Configure Hardware Driver' menu items opens the PCI Audio Console window shown above. This window provides all of the settings for your PCI-424 hardware.

BRIEF OVERVIEW OF PCI-424 SETTINGS

Before you begin using your PCI-424 hardware with Digital Performer, review the settings in the PCI Audio Console window (shown on the previous page in Figure 7-1). The following sections provide a brief explanation of each setting from the perspective of Digital Performer. For a more complete overview of the settings in the MOTU PCI Audio Console window, see chapter 6, "MOTU PCI Audio Console" (page 65).

Sample rate

Choose the desired overall sample rate for the PCI-424 system and Digital Performer. Newly recorded audio in Digital Performer will have this sample rate. Imported audio or soundbites in existing files that do not match this sample rate will be displayed in the Soundbites window with a red 'X' on its move handle to indicate that it cannot be played.

Clock Source

This setting is very important because it determines which audio clock the PCI-424 system will follow.

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

If you are slaving the PCI-424 system to the ADAT sync Input connector on the PCI-424 card, choose *PCI-424: ADAT.*

If you have multiple digital audio devices connected, or if you are not sure about the clock source of your setup, be sure to read "Make sync connections" on page 35 and "Clock Source" on page 66.

Monitor Outputs

The Monitor Outputs option determines which outputs the MOTU PCI Sound Manager driver will use.

Samples Per Buffer

The *Samples Per Buffer* setting can be used to reduce the delay — or *latency* — that you hear when live audio is patched through plug-ins in Digital Performer. 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 Digital Performer. 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.

Smaller settings reduce latency, but they also put more demand on your computer' processor (as shown in the Performance Monitor window). For best results, try 256 samples or lower, which reduces latency enough to be inaudible for most audio material. Keep an eye on the Processor meter in the Performance monitor when making this adjustment. Be sure to try playing some audio, too, as the processor meter can jump up significantly during playback at lower settings. Also, use CueMix Console; it's message center feature helps you find the optimum Samples Per Buffer setting for your system.

For further information about CueMix Console and reducing latency, see chapter 12, "Reducing Monitoring Latency" (page 105).

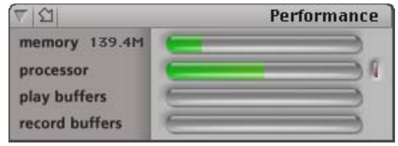


Figure 7-2: When adjusting the 'Samples Per Buffer' to reduce patch thru latency, watch the 'processor' meter in the Performance Monitor. From the Windows menu, choose Performance Monitor. If you hear distortion, or if the Performance meter is peaking, open CueMix Console and look at its message center for hints as to what's going on.

ENABLING INPUTS AND OUTPUTS

The middle portion of the PCI Audio Console window determines which inputs and outputs you'll see in all of Digital Performer's audio input and output menus. To make an input or output available in Digital Performer, check its box. For further information about making these settings, see "Interface settings" on page 70.

BE SURE YOU HAVE ENOUGH VOICES

Once you've enabled the desired audio inputs and outputs on your audio interface(s), go to the Basics menu and choose *MOTU Audio System Options>Configure Studio Size.* Then check to make sure you have enough mono and stereo audio voices for the work you need to do. Consult the *MOTU Audio System* chapter in your Digital Performer manual for further information about the settings in this dialog.

24-BIT OPERATION

Your PCI-424 hardware fully supports Digital Performer's 24-bit recording capabilities, including both analog and digital 24-bit recording. If you would like to record and play back 24-bit audio files, go to the Basics menu, choose *Configure Audio System>Configure Sample Format*, and choose 24-bit recording as the sample format. This setting is saved with the Digital Performer project.

INPUT MONITORING MODE

To determine how Digital Performer will monitor input from the PCI-424 system, go to the Basics menu and choose *Configure Audio System>Input Monitoring Mode*. For complete details, see your Digital Performer manual.

FINE-TUNING I/O TIMING

The PCI-424 system has the ability to be sample accurate. This means that when you transfer audio between Digital Performer and an ADAT or other ADAT Sync-compatible recorder, you can record the audio back and forth as many times as you want between them and it will remain exactly at its original sample location (unless you move it in Digital Performer, of course).

Under normal operating conditions, Digital Performer's timing will be sample-accurate with ADAT Sync-compatible recorders. It will also be sample-accurate with Tascam sync-compatible recorders if you are synchronizing them to a 2408mk3 interface with a Digital Timepiece.

Occasionally, you may encounter a situation in which you observe a slight offset of one sample — or maybe a few — caused by inherent latencies in the devices you are using with the 2408mk3. Usually, these offsets will be consistent, and you can compensate for them in Digital Performer. To do so, choose *Configure Audio System>Fine-tune Audio I/O Timing* from the Basics menu as shown in Figure 7-3.

Figure 7-3: Fine-tuning the timing of audio playback and recording.

SYNCHRONIZATION

Digital Performer can run under its own transport control or slave to an external sync source, such as SMPTE time code or ADAT sync (sample address).

Running DP under its own transport control

If you do not need to synchronize Digital Performer with time code or another recording device, such as a tape deck, just leave the *Slave to External Sync* command in the Basics menu unchecked.

However, even though Digital Performer is not slaving to external sync, you still need to be concerned with the synchronization of the PCI-424 system's digital audio clock with other devices connected to it digitally (if any). For example, if you have a digital mixer connected to a 2408mk3 interface via an ADAT optical lightpipe cable, you need to make sure that their audio clocks are phase-locked. For details, see "Syncing ADAT 'lightpipe' devices" on page 46 and "Make sync connections" on page 35. If you don't have any digital audio devices connected, digital audio phase-lock does not apply to you.

Resolving DP and the PCI-424 to video and/or SMPTE time code

If you need to slave Digital Performer and the PCI-424 system to video and/or SMPTE time code, you can do so with or without a dedicated synchronizer.

Resolving directly to video and/or time code (with no synchronizer)

To resolve your Digital Performer/PCI-424 system directly to video and/or SMPTE time code with no additional synchronization devices, use the setup shown in "Syncing to SMPTE time code" on page 41 or "Syncing to video" on page 42.

☛ Direct video sync requires a 2408mk3 interface. SMPTE time code sync can be done with any MOTU PCI audio interface.

Choose *Receive Sync* from the Basics menu and choose the *Sample accurate* option. Then make sure that the *Slave to External Sync* command in the Basic menu is checked. Make sure the *Clock Source* setting in the MOTU PCI Audio Console window is set to *PCI-424: SMPTE* or *PCI-424: Video*. Also, make sure that you've specified the SMPTE time code input (if necessary) in the MOTU SMPTE Console applet, as shown in Figure 3-23 on page 41 or Figure 3-24 on page 42.

Resolving to video and/or time code with a dedicated synchronizer

To resolve your Digital Performer/PCI-424 system to video and/or SMPTE time code using an additional synchronization device, use the setup shown in "Syncing to video and/or SMPTE time code using a synchronizer" on page 43.

Choose *Receive Sync* from the Basics menu and choose the MTC (MIDI Time Code) option. Then make sure that the *Slave to External Sync* command in the Basic menu is checked. To ensure that your audio tracks don't drift out of sync with your MIDI tracks — or time code, use a hardware synchronizer like the MIDI Timepiece AV or Digital Timepiece to resolve the PCI-424 hardware as well, as shown in Figure 3-25 on page 43. A digital audio synchronizer is required for drift-free SMPTE/MIDI time code sync. Make sure the *Clock Source* setting in the MOTU PCI Audio Console window has the appropriate setting for locking the PCI-424 system to the synchronizer. For example, in Figure 3-25 on page 43, word clock is being used to resolve a 2408mk3, so the Clock Source setting is *2408mk3: Word Clock In*.

☛ If you have an ADAT sync or a Tascam sync compatible device, don't use SMPTE time code. Instead, use sample-accurate sync as described in the next section.

Sample-accurate sync

Together, Digital Performer and the PCI-424 system provide you with sample-accurate transfers with ADATs, Alesis recorders, or any other devices that support standard ADAT sample address (*ADAT Sync*).

Similarly, with the help of a MOTU Digital Timepiece, Digital Performer and a 2408mk3 system can perform sample-accurate transfers with Tascam digital recorders.

A sample-accurate transfer is one in which the original location of the audio is preserved in the transfer, down to the sample.

For details on how to set up sample-accurate sync, see "Sample-accurate sync" on page 37. Be sure to choose the *Sample Accurate Sync* option in Digital Performer's *Receive Sync* dialog (Basics menu), and make sure that the *Slave to External Sync* command is checked in the Basic menu, too.

To control the transports of everything together from Digital Performer, see the next section.

MIDI MACHINE CONTROL (MMC)

If you have ADATs and a MMC-compatible ADAT synchronizer like the MIDI Timepiece AV and Digital Timepiece, you can control everything from your computer screen with Digital Performer's transport controls and cueing features (like Markers, the playback wiper, etc.)

Similarly, if you have Tascam recorders and a MOTU Digital Timepiece (or other MMC-compatible Tascam synchronizer), can control all of your Tascam decks (in ABS time) in a similar fashion from Digital Performer.

See the MIDI Machine Control chapter in your MIDI Timepiece AV or Digital Timepiece manual for details on how to set this up.

REDUCING DELAY WHEN PROCESSING LIVE INPUTS WITH PLUG-INS

If you patch a live input (such as MIDI synthesizer) through a plug-in effect in Digital Performer, you might hear a slight delay. There are several ways to reduce this delay. For details, see chapter 12, “Reducing Monitoring Latency” (page 105).

EXCHANGING PROJECTS WITH AUDIODESK

Digital Performer (Version 2.6 or later) can exchange files with AudioDesk 1.0. For example, you can transfer a file from Digital Performer to AudioDesk, and back again. Just use *Save As* in Digital Performer’s File menu and choose the *AudioDesk* file format. To open AudioDesk files in Digital Performer, just use the Open command. (No conversion is required beforehand in AudioDesk.)

If you have an earlier version of Digital Performer (2.5 or earlier), you can open your Digital Performer files in AudioDesk (with the Open command in the File menu), but Digital Performer 2.5 or earlier cannot open AudioDesk files.

MONITORING SYSTEM PERFORMANCE

Because it has so many inputs and outputs, a PCI-424 system pushes the limits of your computer’s processing power. Keep the Audio Performance window open (from the Audio menu) to keep tabs on the load on your CPU and disk buffers. If the meters get too high, you can reduce the load by reducing the number of inputs and outputs you are working with. Use the MOTU PCI Audio Console to uncheck input check boxes and set output source menus to *None*.

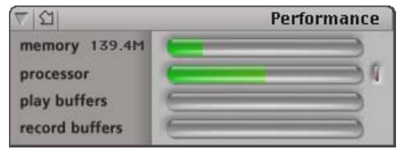


Figure 7-4: Keep the Audio Performance window open to keep tabs on your computer’s processing power and hard disk performance.

SOUND MANAGER AND DIGITAL PERFORMER

Digital Performer includes a MOTU Audio System plug-in called AudioTap that allows you to route any Sound Manager audio into Digital Performer’s mixing environment. From there, you can route it to your 2408mk3 interface via any of Digital Performer’s extensive audio routing features. For details, consult your Digital Performer documentation.

CHAPTER 8 AudioDesk

OVERVIEW

This chapter provides a brief overview of AudioDesk's basic I/O and synchronization operation with the PCI-424 system. For complete information about all of AudioDesk's powerful workstation features, see the AudioDesk manual included with your PCI-424 system.

SETTING UP YOUR SYSTEM

As described in chapter 4, "Installing the PCI-424 Macintosh Software" (page 53), the AudioDesk and PCI-424 software installer will properly install and update all of the necessary software components in your Macintosh system. For your convenience, here is a brief summary of what AudioDesk requires:

- The MOTU PCI Audio Driver must be in the MOTU folder (as shown in Figure 4-1 on page 54).
- The MOTU Audio System item in the MOTU Folder (as shown in Figure 4-1 on page 54) must be the same version as that which gets installed by the PCI-424 installer CD-ROM (or later).
- If you will be using AudioDesk's MIDI Machine Control (MMC) or MIDI Time Code sync features, FreeMIDI must be installed. (You can do this from the PCI-424 installer CD.)
- If you are using a MIDI Timepiece AV or Digital Timepiece for synchronization, be sure they are present in your FreeMIDI setup.

THE MOTU PCI AUDIO CONSOLE WINDOW

The MOTU PCI Audio Console window gives you direct access to all of the settings in your PCI-424 system hardware. To open it, choose *MOTU Audio System options>Configure Hardware Driver* from the Basics menu and choose *MOTU PCI Audio* from the menu at the top of the Configure Hardware dialog.

The PCI Audio Console window is central to the operation of your PCI-424 hardware with AudioDesk. For complete details about the important settings in this window, see chapter 6, "MOTU PCI Audio Console" (page 65).

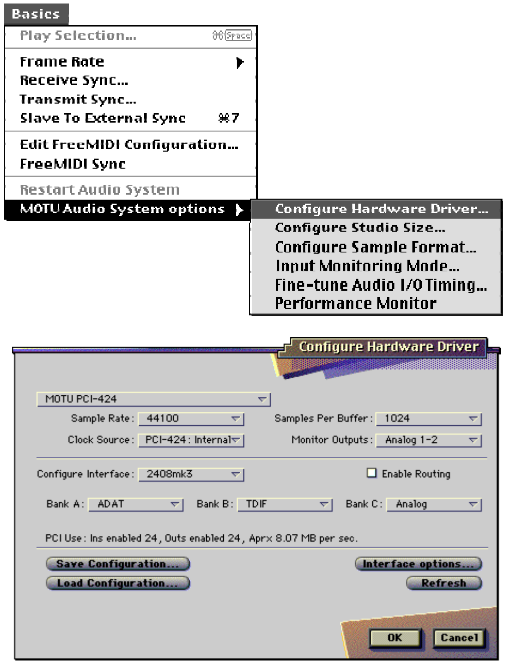


Figure 8-1: The 'Configure Hardware Driver' menu items opens the PCI Audio Console window shown above. This window provides all of the settings for your PCI-424 hardware.

BRIEF OVERVIEW OF PCI-424 SETTINGS

Before you begin using your PCI-424 hardware with AudioDesk, review the settings in the PCI Audio Console window (shown in Figure 8-1). The following sections provide a brief explanation of each setting from the perspective of AudioDesk. For a more complete overview of the settings in the MOTU PCI Audio Console window, see chapter 6, "MOTU PCI Audio Console" (page 65).

Sample rate

Choose the desired overall sample rate for the PCI-424 system and AudioDesk. Newly recorded audio in AudioDesk will have this sample rate. Imported audio or soundbites in existing files that do not match this sample rate will be displayed in the Soundbites window with a red 'X' on its move handle to indicate that it cannot be played.

Clock Source

This setting is very important because it determines which audio clock the PCI-424 system will follow.

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

If you are slaving the PCI-424 system to the ADAT sync Input connector on the PCI-424 card, choose *PCI-424: ADAT.*

If you have multiple digital audio devices connected, or if you are not sure about the clock source of your setup, be sure to read "Make sync connections" on page 35 and "Clock Source" on page 66.

Samples Per Buffer

The *Samples Per Buffer* setting can be used to reduce the delay — or *latency* — that you hear when live audio is patched through plug-ins in AudioDesk. 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 AudioDesk. 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.

Smaller settings reduce latency, but they also put more demand on your computer' processor (as shown in the Performance Monitor window). For best results, try 256 samples or lower, which reduces latency enough to be inaudible for most audio material. Keep an eye on the Processor meter in the Performance monitor when making this adjustment. Be sure to try playing some audio, too, as the processor meter can jump up significantly during playback at lower settings. Also, use CueMix Console; it's message center feature helps you find the optimum Samples Per Buffer setting for your system.

For further information about CueMix console and reducing latency, see chapter 12, "Reducing Monitoring Latency" (page 105).

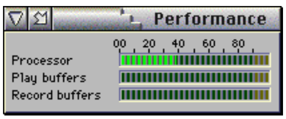


Figure 8-2: When adjusting the 'Samples Per Buffer' to reduce patch thru latency, watch the 'processor' meter in the Performance Monitor. From the Basics menu, choose 'MOTU Audio System options>Performance Monitor'. If you hear distortion, or if the Performance meter is peaking, open CueMix Console and look at its message center for hints as to what's going on.

Monitor Outputs

This setting determines where you'll hear things like soundbite auditioning, audio scrubbing, and real-time previewing of Adobe Premiere audio plug-ins.

ENABLING INPUTS AND OUTPUTS

The middle portion of the PCI Audio Console window determines which inputs and outputs you'll seen in all of AudioDesk's audio input and output menus. To make an input or output available in AudioDesk, check its box. For further information about making these settings, see "Interface settings" on page 70.

BE SURE YOU HAVE ENOUGH VOICES

Once you've enabled the desired PCI-424 audio inputs and outputs, go to the Basics menu and choose *MOTU Audio System Options>Configure Studio Size*. Then check to make sure you have enough mono and stereo audio voices for the work you need to do. Consult the *MOTU Audio System* chapter in your AudioDesk manual for further information about the settings in this dialog.

24-BIT OPERATION

Your PCI-424 hardware fully supports AudioDesk's 24-bit recording capabilities, including both analog and digital 24-bit recording. If you would like to record and play back 24-bit audio files, go to the Basics menu, choose *MOTU Audio System options>Configure Sample Format*, and choose 24-bit recording as the sample format. This setting is saved with the AudioDesk project.

INPUT MONITORING MODE

To determine how AudioDesk will monitor input from the PCI-424 system, go to the Basics menu and choose *MOTU Audio System options>Input Monitoring Mode*. For complete details, see your AudioDesk manual.

SYNCHRONIZATION

AudioDesk can run under its own transport control or slave to an external sync source, such as SMPTE time code or ADAT sync (sample address).

Running AudioDesk under its own transport control

If you do not need to synchronize AudioDesk with time code or another recording device, such as a tape deck, just leave the *Slave to External Sync* command in the Basics menu unchecked.

However, even though AudioDesk is not slaving to external sync, you still need to be concerned with the synchronization of the PCI-424 system's digital audio clock with other devices connected to it digitally (if any). For example, if you have a digital mixer connected to a 2408mk3 interface via an ADAT optical lightpipe cable, you need to make sure that their audio clocks are phase-locked. For details, see "Syncing ADAT 'lightpipe' devices" on page 46 and "Make sync connections" on page 35.

If you don't have any digital audio devices connected, digital audio phase-lock does not apply to you.

Resolving AudioDesk and the PCI-424 to video and/or SMPTE time code

If you need to slave AudioDesk and the PCI-424 system to video and/or SMPTE time code, you can do so with or without a dedicated synchronizer.

Resolving directly to video and/or time code (with no synchronizer)

To resolve your AudioDesk/PCI-424 system directly to video and/or SMPTE time code with no additional synchronization devices, use the setup shown in "Syncing to SMPTE time code" on page 41 or "Syncing to video" on page 42.

☛ Direct video sync requires a 2408mk3 interface. SMPTE time code sync can be done with any MOTU PCI audio interface.

Choose *Receive Sync* from the Basics menu and choose the *Sample accurate* option. Then make sure that the *Slave to External Sync* command in the Basic menu is checked. Make sure the *Clock Source* setting in the MOTU PCI Audio Console window is set to *PCI-424: SMPTE* or *PCI-424: Video*. Also, make sure that you've specified the SMPTE time code input (if necessary) in the MOTU SMPTE Console applet, as shown in Figure 3-23 on page 41 or Figure 3-24 on page 42.

Resolving to video and/or time code with a dedicated synchronizer

To resolve your AudioDesk/PCI-424 system to video and/or SMPTE time code using an additional synchronization device, use the setup shown in "Syncing to video and/or SMPTE time code using a synchronizer" on page 43.

Choose *Receive Sync* from the Basics menu and choose the MTC (MIDI Time Code) option. Then make sure that the *Slave to External Sync* command in the Basic menu is checked. To ensure that your audio tracks don't drift out of sync with your MIDI tracks — or time code, use a hardware synchronizer like the MIDI Timepiece AV or Digital Timepiece to resolve the PCI-424 hardware as well, as shown in Figure 3-25 on page 43. A digital audio synchronizer is required for drift-free SMPTE/MIDI time code sync. Make sure the *Clock Source* setting in the MOTU PCI Audio Console window has the appropriate setting for locking the PCI-424 system to the synchronizer. For example, in Figure 3-25 on page 43, word clock is being used to resolve a 2408mk3, so the Clock Source setting is *2408mk3: Word Clock In*.

☛ If you have an ADAT sync or a Tascam sync compatible device, don't use SMPTE time code. Instead, use sample-accurate sync as described in the next section.

Sample-accurate sync

Together, AudioDesk and the 2408mk3 system provide you with sample-accurate transfers with ADATs, Alesis recorders or any other devices that support standard ADAT sample address (*ADAT Sync*).

Similarly, with the help of a MOTU Digital Timepiece, AudioDesk and a 2408mk3 system can perform sample-accurate transfers with Tascam digital recorders.

A sample-accurate transfer is one in which the original location of the audio is preserved in the transfer, down to the sample.

For details on how to set up sample-accurate sync, see "Sample-accurate sync" on page 37. Be sure to choose the *Sample Accurate Sync via PCI-424* option in AudioDesk's *Receive Sync* dialog (Basics menu), and make sure that the *Slave to External Sync* command is checked in the Basic menu, too.

To control the transports of everything together from AudioDesk, see the next section.

MIDI MACHINE CONTROL (MMC)

If you have ADATs and a MMC-compatible ADAT synchronizer like the MIDI Timepiece AV and Digital Timepiece, you can control everything from your computer screen with AudioDesk's transport controls and cueing features (like Markers, the playback wiper, etc.)

Similarly, if you have Tascam recorders and a MOTU Digital Timepiece (or other MMC-compatible Tascam synchronizer), can control all of your Tascam decks (in ABS time) in a similar fashion from AudioDesk.

See the MIDI Machine Control chapter in your MIDI Timepiece AV or Digital Timepiece manual for details on how to set this up.

REDUCING DELAY WHEN PROCESSING LIVE INPUTS WITH PLUG-INS

If you patch a live input (such as MIDI synthesizer) through a plug-in effect in AudioDesk, you might hear a slight delay. There are several ways to reduce this delay. For details, see "Samples Per Buffer" on page 84.

EXCHANGING PROJECTS WITH DIGITAL PERFORMER

AudioDesk 1.0 can exchange files with Digital Performer (Version 2.6 or later). For example, you can transfer a file from Digital Performer to AudioDesk, and back again. Just use *Save As* in Digital Performer's File menu and choose the *AudioDesk* file format. To open AudioDesk files in Digital Performer, just use the Open command. (No conversion is required beforehand in AudioDesk.)

If you have an earlier version of Digital Performer (2.5 or earlier), you can open your Digital Performer files in AudioDesk (with the Open command in the File menu), but Digital Performer 2.5 or earlier cannot open AudioDesk files.

MONITORING SYSTEM PERFORMANCE

Because it has so many inputs and outputs, a PCI-424 system pushes the limits of your computer's processing power. Keep the Audio Performance window open (from the Audio menu) to keep tabs on the load on your CPU and disk buffers. If the meters get too high, you can reduce the load by reducing the number of inputs and outputs you are working with. Use the MOTU PCI Audio Console to uncheck input check boxes and set output source menus to *None*.

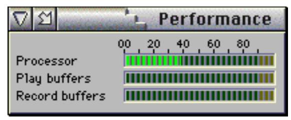


Figure 8-3: Keep the Audio Performance window open to keep tabs on your computer's processing power and hard disk performance.

AUDIODESK AND MIDI SEQUENCING

AudioDesk can play audio as a background application, allowing you to run a sequencer at the same time in the foreground. However, there is no way to continuously synchronize — or *resolve* — a sequencer with AudioDesk, so the two programs will eventually drift out of sync, even if you manage to start them at the same time. If you'd like to do integrated MIDI sequencing, your best bet is Digital Performer, which offers all of the same features as AudioDesk (and a whole lot more), along with powerful, state-of-the-art MIDI sequencing. Talk to your authorized MOTU dealer for details about upgrading from AudioDesk to Digital Performer.

CHAPTER 9 Cubase, Nuendo and other ASIO Software

OVERVIEW

The PCI-424 system 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 VST and Nuendo.

☛ Note to Nuendo Macintosh users: the examples in this chapter show screen shots of Nuendo for Windows, but they are very similar to the Mac version.

PREPARATION

Before you run Cubase with your PCI-424 system, launch AudioDesk and play back the demo project to make sure that the PCI-424 hardware software drivers are set up properly. The AudioDesk demo project is located on the PCI-424 Installer CD. Drag it to your hard drive before opening it in AudioDesk, as your CD drive will be too slow to play the audio.

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, "Installing the PCI-424 Hardware" (page 25).
- chapter 4, "Installing the PCI-424 Macintosh Software" (page 53)
- chapter 8, "AudioDesk" (page 83)

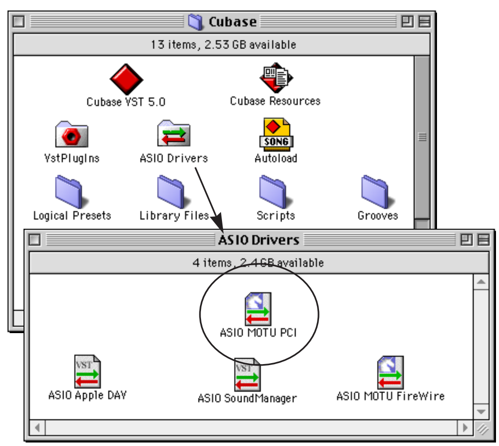


Figure 9-1: The PCI-424 installer puts the PCI-424 ASIO driver in the Cubase ASIO Drivers folder.

RUN MOTU PCI AUDIO CONSOLE BEFORE CUBASE

Before you run Cubase, launch the MOTU PCI Audio Console to configure your PCI-424 hardware. The MOTU PCI 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 the MOTU PCI Audio Console, see chapter 6, "MOTU PCI Audio Console" (page 65).

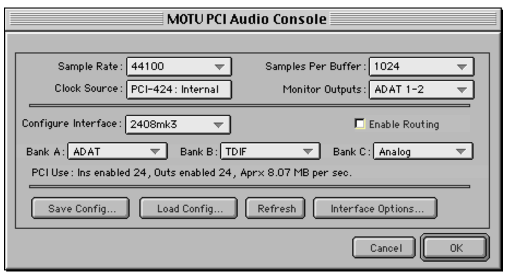


Figure 9-2: This example shows a 2408mk3 interface. Configure the three banks as you would like to use them in Cubase. Only activate the banks you really need, though, to conserve CPU and PCI bandwidth.

☛ Changing input and output settings in the MOTU PCI Audio Console causes Cubase to lose the input and bussing assignments in any files that don't match the new settings. To avoid this, you can save a PCI-424 setup on disk with your Cubase file. Then, the next time you open the Cubase file, you can load the 424 configuration first so that no settings are lost. For details, see "Saving and recalling routings" on page 75.

CHOOSING THE PCI-424 DRIVER IN CUBASE

Once you've made the preparations described so far in this chapter, you're ready to run Cubase.

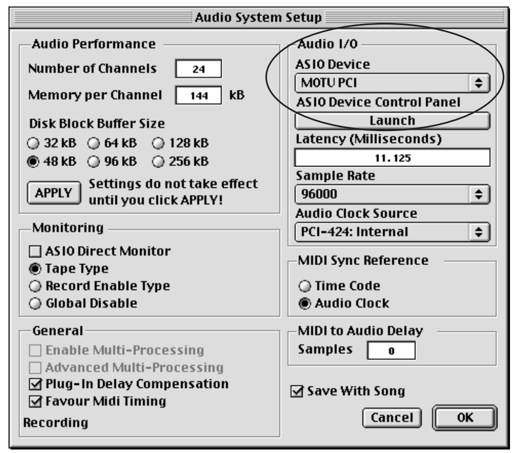
Cubase VST

To activate the PCI-424 driver in Cubase VST, choose *Audio Setup>System* from the Options menu, and then choose *MOTU PCI* from the ASIO device menu. Make the other settings in the dialog as need for your system and synchronization scenario.

Nuendo

To activate the PCI-424 driver in Nuendo or , go to the Device Setup window, click VST Multitrack and choose MOTU PCI ASIO from the ASIO Driver menu as shown below. Make the other settings in the dialog as need for your system and synchronization scenario.

Cubase VST



Nuendo

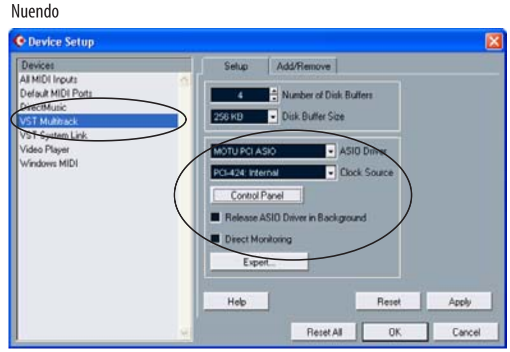


Figure 9-3: Activating the PCI-424 driver in Nuendo and Cubase.

NUMBER OF CHANNELS

In Cubase VST, be sure to choose enough channels in the System dialog (as shown above in Figure 9-3) to cover the number of inputs and outputs provided by your PCI-424 system.

THE LAUNCH BUTTON

The Mac version of Cubase does not allow the MOTU PCI Audio Console to run at the same time as Cubase. Therefore, the Launch button in the System dialog as shown in Figure 9-3 will not launch the MOTU PCI Audio Console. Perhaps this will be addressed in a future update of Cubase. In the meantime, you can access the MOTU PCI Audio Console in one of two ways:

- Quit Cubase, and then run the MOTU PCI Audio Console from the Finder, OR
- Temporarily switch to a different ASIO Device in the System dialog, and then run the MOTU PCI Audio Console from the Finder

In either case, any changes you make to the MOTU PCI Audio Console window will be reflected in Cubase when you reactivate the MOTU PCI-424 ASIO driver in Cubase.

REDUCING DELAY WHEN PROCESSING LIVE INPUTS WITH 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 12, "Reducing Monitoring Latency" (page 105).

ASIO DIRECT MONITORING

The ASIO Direct Monitoring option (Figure 9-3) allows you to monitor inputs directly in the hardware with no drain on your computer and near zero latency. When you enable this option, Cubase uses the PCI-424's CueMix monitoring features whenever you use Cubase's monitoring features. For further information, see "Controlling CueMix DSP from within Cubase" on page 110.

AUDIO CLOCK SOURCE

This setting is the same as the Clock Source setting in the MOTU PCI Audio Console window. It determines which audio clock the PCI-424 system will slave to. Choose the setting that is most appropriate for your synchronization scenario. For complete details, see "Clock Source" on page 66.

OTHER SYSTEM DIALOG SETTINGS

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

ACTIVATING PCI-424 INPUTS

Once you've chosen the MOTU PCI-424 ASIO driver in the Audio System dialog, choose *VST Inputs* from the *Panels* menu (or the *Devices* menu in Cubase SX) to see the PCI-424 inputs. To activate them, click the *Active* light next to each input.

The inputs that appear in the VST Inputs window correspond to the banks you enabled (or individual inputs you checked) in the MOTU PCI Audio Console, as shown in Figure 9-2 on page 90.

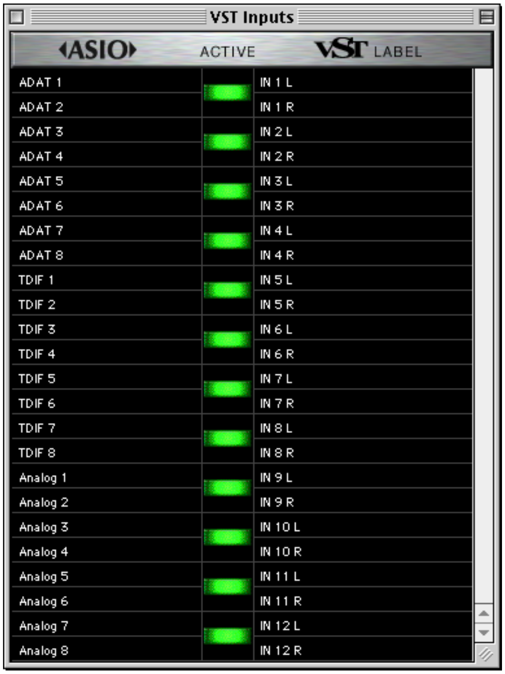


Figure 9-4: Activating PCI-424 inputs in Cubase VST.

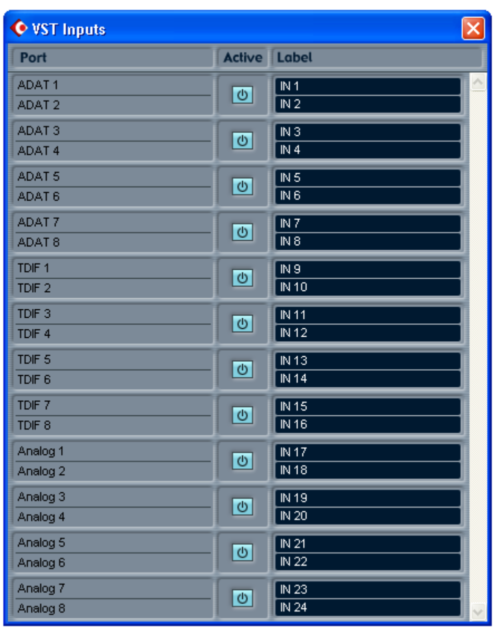


Figure 9-5: Activating PCI-424 inputs in Nuendo.

ASSIGNING INPUTS

Once you've activated the PCI-424 system inputs as shown in the previous section, you can then assign them to Cubase or Nuendo audio channels in the channel mixers in the usual fashion.

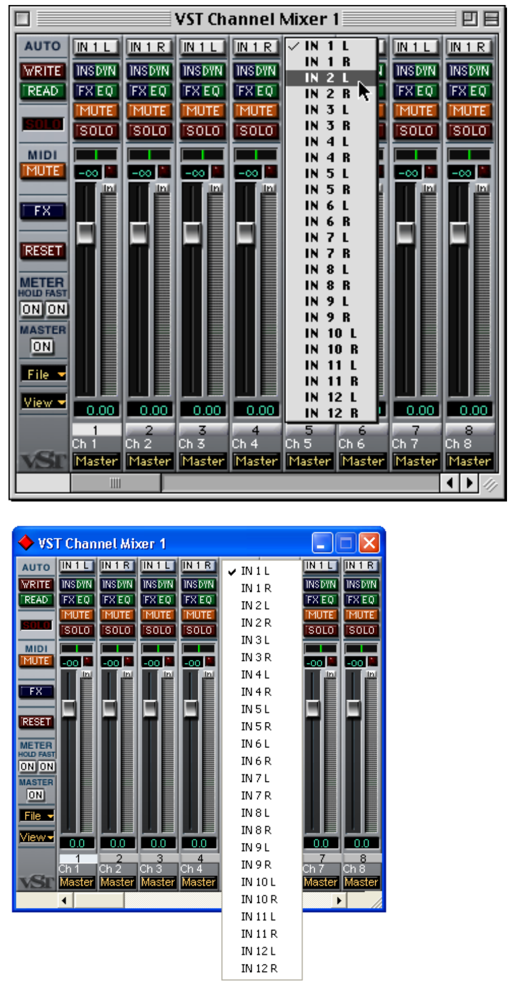


Figure 9-6: To assign a PCI-424 input to a Cubase VST audio channel: command-click the input button at the top of the channel strip. For Nuendo or , consult your documentation.

ASSIGNING OUTPUTS

As shown earlier in Figure 9-2 on page 90, any banks that you have enabled in the MOTU PCI Audio Console will be available in Cubase or Nuendo as outputs. In Cubase VST, these outputs appear in the VST Master Mixer window as output assignments for the master fader and busses, as shown below in Figure 9-7.In Nuendo, they appear in the VST Outputs window.

CHANGING PCI-424 SETTINGS

To change the PCI-424 settings at any time, run the MOTU PCI Audio Console. See “The Launch button” on page 91 for details. In Nuendo, go to the Device Setup window and click the ASIO Control Panel button, as shown in Figure 9-3 on page 90.

In Cubase VST, use the output buttons at the bottom of each channel strip, including the master fader, to assign PCI-424 outputs to busses. You can then assign channels in the VST Master Mixer window to each bus as desired.

In Nuendo, access the PCI-424 outputs via the busses in the VST Outputs window.

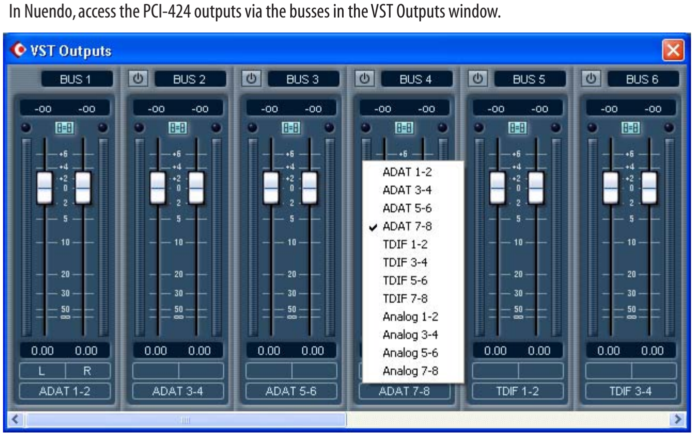


Figure 9-7: Working with PCI-424 outputs in Nuendo or Cubase.

SYNCHRONIZATION

Cubase or Nuendo can run under its own transport control or slave to SMPTE time code. It can also perform sample-accurate digital audio transfers with Alesis digital recorders and Tascam family digital recorders.

As you read through the following sections to decide what form of synchronization you might need with other devices in your studio, be sure to consult chapter 3, "Installing the PCI-424 Hardware" (page 25) for the proper hardware connections. Use the synchronization diagrams in that chapter to be clear about how you will be synchronizing Cubase to the other components of your system.

Running Cubase or Nuendo under its own transport control

If you do not need to synchronize Cubase or Nuendo with time code or another recording device, such as a tape deck, just leave its SMPTE time code synchronization features disabled.

However, even though Cubase or Nuendo is not slaving to SMPTE time code, you still need to be concerned with the synchronization of the PCI-424 system's digital audio clock with other devices connected to it digitally (if any). For example, if you have a digital mixer connected to a 2408mk3 interface via an ADAT optical lightpipe cable, you need to make sure that their audio clocks are phase-locked. For details, see "Syncing ADAT 'lightpipe' devices" on page 46 and "Make sync connections" on page 35. If you don't have any digital audio devices connected, digital audio phase-lock does not apply to you.

Resolving Cubase or Nuendo and the PCI-424 to video and/or SMPTE time code

If you need to slave Cubase or Nuendo and the PCI-424 system to video and/or SMPTE time code, you can do so with or without a dedicated synchronizer. SMPTE time code synchronization can be accomplished via the analog input of any connected interface. Direct video synchronization requires as 2408mk3 interface.

Resolving directly to video and/or time code (with no synchronizer)

To resolve your PCI-424 system directly to video and/or SMPTE time code with no additional synchronization devices, use the setup shown in "Syncing to SMPTE time code" on page 41 or "Syncing to video" on page 42.

☛ Direct video sync requires a 2408mk3 interface. SMPTE time code sync can be done with any MOTU PCI audio interface.

First, set up Cubase or Nuendo for sample-accurate sync as explained in "Sample-accurate sync with Cubase or Nuendo" on page 95. Make sure the *Clock Source* setting in the MOTU PCI Audio Console window is set to *PCI-424: SMPTE* or *PCI-424: Video*. Also, make sure that you've specified the SMPTE time code input (if necessary) in the MOTU SMPTE Console applet, as shown in Figure 3-23 on page 41 or Figure 3-24 on page 42.

Resolving to video and/or time code with a dedicated synchronizer

To resolve your PCI-424 system to video and/or SMPTE time code using an additional synchronization device, use the setup shown in "Syncing to video and/or SMPTE time code using a synchronizer" on page 43.

Follow the instructions in your Cubase or Nuendo manual for slaving them to MIDI Time Code (MTC). To ensure that your audio tracks don't drift out of sync with your MIDI tracks — or time code, use a hardware synchronizer like the MIDI Timepiece AV or Digital Timepiece to resolve the PCI-424 hardware as well, as explained in "Syncing to video and/or SMPTE time code using a synchronizer" on page 43. A digital audio synchronizer is required for drift-free SMPTE/MIDI time code sync. Make sure the *Clock Source* setting in the MOTU PCI Audio Console window has the appropriate setting for locking the PCI-424 system to the synchronizer. For example, in Figure 3-25 on page 43, word clock is being used to resolve a 2408mk3 interface, so the Clock Source setting is *2408mk3: Word Clock In*.

☛ If you have an ADAT sync or a Tascam sync compatible device, don't use SMPTE time code. Instead, use sample-accurate sync as described in the next section.

Sample-accurate sync with Cubase or Nuendo

Cubase and Nuendo, along with the PCI-424 system and its ASIO 2 driver, provide you with sample-accurate transfers with ADATs and any other devices that support standard ADAT sample address (*ADAT Sync*).

Similarly, with the help of a MOTU Digital Timepiece universal A/V synchronizer, Cubase (or Nuendo) and a 2408mk3-based PCI-424 system can perform sample-accurate transfers with Tascam digital recorders.

A sample-accurate transfer is one in which the original location of the audio is preserved in the transfer, down to the sample.

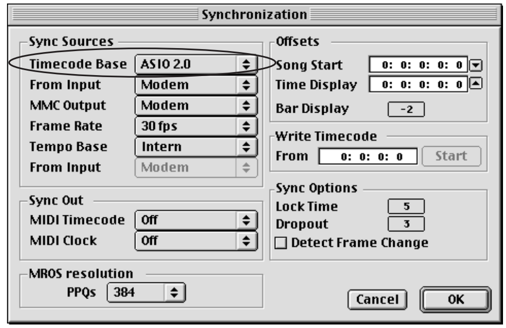
For details on how to connect your hardware for sample-accurate sync, see "Sample-accurate sync" on page 37. Then, set up Cubase as follows:

☛ Before you begin, in Cubase's MIDI System Setup window, set OMS compatibility to *No OMS*. Cubase does not appear to be able to achieve sample-accurate sync when running under OMS.

1 Choose *PCI-424: ADAT* as the *Audio Clock Source* setting. In Cubase VST, this setting is in the Audio System Setup window (Audio menu). In Nuendo, this setting is in the Device Setup window (Options menu).

2 Go to Cubase or Nuendo's *Synchronization* window, as shown below:

Cubase VST



Nuendo

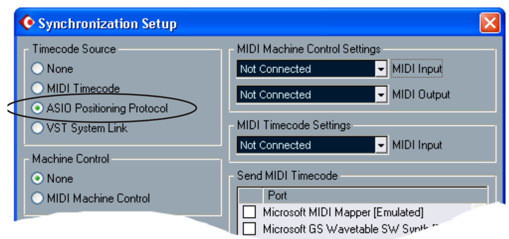


Figure 9-8: Setting up sample-accurate sync via ASIO 2.

3 If you are *not* using an MMC-compatible synchronizer (such as a MOTU MIDI Timepiece AV, Digital Timepiece or Alesis BRC), choose the settings shown above in Figure 9-8 that applies to you. In this scenario, transport control is handled by the ADAT (or other sample-accurate sync source).

4 If you *are* using an MMC-compatible synchronizer (such as a MOTU MIDI Timepiece AV, Digital Timepiece or Alesis BRC), set Cubase VST's *Sync Source Timecode Base* to *ASIO 2.0 MMC* or enable Nuendo's *MIDI Machine Control* option. In addition, choose the appropriate MIDI port for the MMC synchronizer from VST's *Output* menu or Nuendo's MIDI machine Control *MIDI Output* menu. If you're using a MIDI Timepiece AV, you can choose any of its MIDI ports in this menu. Doing so makes Cubase or

Nuendo send the MMC control messages to the MTP AV (or other MMC device). In this scenario, transport control is handled by Cubase or Nuendo itself.

5 In Cubase VST's Controls window, enable SYNC. In Nuendo, enable (check) the *Sync Online* command in the Transport menu.

Figure 9-9: Enabling the SYNC button.

6 Begin playback from the sample-accurate sync source (ADAT, DA-88, etc.) Transport control is handled by the sample-accurate sync source.

MIDI MACHINE CONTROL (MMC)

If you have ADATs (or other ADAT Sync-compatible recorders) and a MMC-compatible ADAT synchronizer like the MIDI Timepiece AV and Digital Timepiece, you can control everything from your computer screen with Cubase's transport controls and cueing features (like the playback wiper, etc.)

Similarly, if you have Tascam digital recorders and a MOTU Digital Timepiece (or other MMC-compatible Tascam synchronizer), you can control all of your Tascam tape decks (in ABS time) in a similar fashion from Cubase.

See "Sample-accurate sync with Cubase or Nuendo" on page 95 for details on how to set this up.

24-BIT OPERATION

Your PCI424 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 PCI-424 system 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 PCI-424 hardware.

MONITORING SYSTEM PERFORMANCE

Because it has so many inputs and outputs, the PCI-424 system pushes the limits of your computer's processing power. Keep the VST Performance window open to keep tabs on the load on your CPU and disk buffers. If the meters get too high, you can reduce the load by reducing the number of inputs and outputs you are working with. Use the MOTU PCI Audio Console to uncheck input check boxes and set output source menus to *None*.

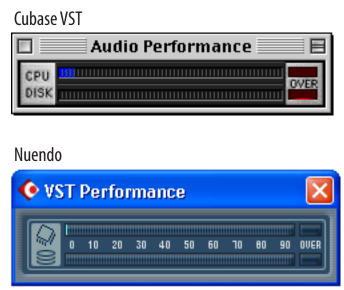


Figure 9-10: Keep the Audio Performance window open to keep tabs on your computer's processing power and hard disk performance.

CHAPTER 10 Sound Manager

OVERVIEW

The Apple *Sound Manager* is a standard Mac OS 9 System Extension that provides stereo sound capabilities on your Power Macintosh.

The PCI-424 system ships with a Sound Manager driver for the PCI-424 audio card. This driver allows your Power Macintosh to use any pair of inputs and outputs (analog or digital) on your PCI-424 system.

Figure 10-1: The Sound Manager driver System Extension for the PCI-424.

USING 3RD PARTY AUDIO SOFTWARE

Just about every dedicated audio application for Macintosh supports Sound Manager, along with many other applications, like video editing software, that handle audio in one form or another. Once you've installed and configured the MOTU PCI Sound Manager driver as described in this chapter, you'll be able to use any pair of 2408mk3 inputs and outputs (analog or digital) to record and play back audio from any Sound Manager compatible program.

SOUND MANAGER IS STEREO ONLY

Sound Manager currently only supports stereo input and output. Therefore, Sound Manager compatible programs can only record two channels at a time with your PCI-424 hardware. Similarly, they can play back audio through one stereo output pair on a PCI-424 audio interface, although most multi-track audio applications can internally mix more than two tracks to stereo. Consult your software's documentation for details.

44.1 AND 48 KHZ ONLY

Sound Manager supports sample rates up to 48 kHz, but nothing higher, so be sure to set the PCI-424 system sample rate to 44.1 or 48 kHz.

INSTALLING THE PCI SOUND MANAGER DRIVER

The MOTU PCI Sound Manager Driver is a standard Macintosh System Extension. When properly installed, it resides in the Extensions Folder of the System Folder on your computer's startup disk.

To install the PCI-424 Sound Manager driver:

1 Insert the PCI-424 installer CD into your Macintosh.

2 Run the PCI-424 software installer.

3 Enable (check) the MOTU PCI Sound Manager Driver option, along with any other packages you would like to install at this time.

4 Click Install.

5 Complete the installation as instructed by the installer. Because the MOTU PCI Sound Manager driver is a System Extension, the installer will ask you to restart the computer.

SOUND CONTROL PANEL IS REQUIRED

The *Sound* Control Panel allows you choose your PCI-424 audio interface as an output for Sound Manager.

Locating the Sound Control Panel

If the *Sound* Control Panel is not present in your Control Panels folder, use Sherlock (command-F) in the Finder to locate it.

If the Sound Control Panel can't be found, insert the Mac OS Installer CD-ROM that shipped with your computer and try again, searching on the CD-ROM. It will probably be there.

Once you've located the Sound Control Panel, you can place it anywhere you like on your hard drive, as it operates like a standard application. Just double-click it to launch it.

INSTALLATION SUMMARY

Here is what you should have by now:

- The *MOTU PCI Sound Manager driver* is installed as a System Extension.
- The *Sound* Control panel is somewhere on your hard drive.
- As part of the standard PCI-424 software installation, a copy of the *MOTU PCI Audio Console* application is now on your hard drive.

If all these things are true, you are ready to activate the PCI-424 hardware as your Sound Manager input source and output destination.

ENABLING INPUTS AND OUTPUTS

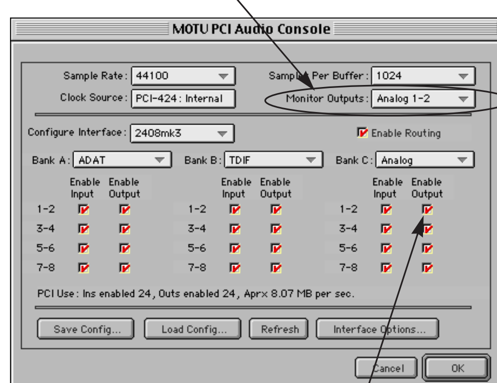
Before you set up Sound Manager, you need to run the MOTU PCI Audio Console and enable the input pair and output pair you would like to use for Sound Manager.

1 Launch the MOTU PCI Audio Console.

2 Choose the desired sample rate.

3 From the *Monitor Outputs* menu, choose the desired output pair for Sound Manager as shown below in Figure 10-2. If the pair you want is not available in the menu, click the *Enable Routing* check box and then click the *Enable Output* check box for the desired output pair.

The 'Monitor Outputs' setting determines which output pair Sound Manager will use. If the pair you want is not available in the menu, click the Enable Routing check box and then click the Enable Output check box for the desired output pair.



Check the output pair you want to use for Sound Manager.

Figure 10-2: In this example, analog outputs 1-2 on a 2408mk3 interface will be used for Sound Manager output.

4 As shown in Figure 10-2, check the box next to the input pair you would like to use for Sound Manager.

5 Quit the MOTU PCI Audio Console.

CONFIGURING THE SOUND CONTROL PANEL

Once you've made the preparations discussed in the previous section in the MOTU PCI Audio Console, you are ready to make your final settings in the Sound Control Panel as follows:

1 Open the Sound Control Panel.

2 Choose *Alert Sounds* from the menu.

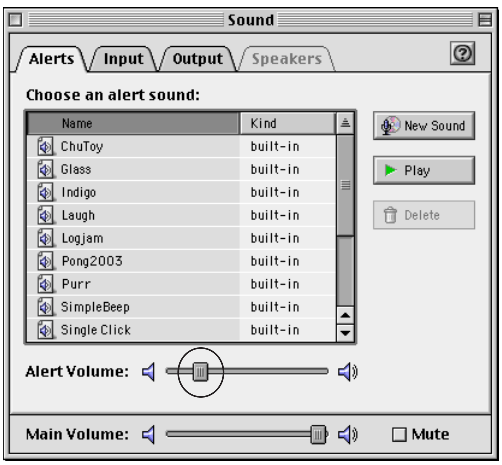


Figure 10-3: Lower the alert sound volume to avoid getting blasted by it through your PCI-424 hardware.

3 Lower the *Alert Volume*. This prevents the Macintosh alert sound from blurting out of your 2408mk3 system.

4 Click the *Input* tab and choose the desired 2408mk3 input pair.

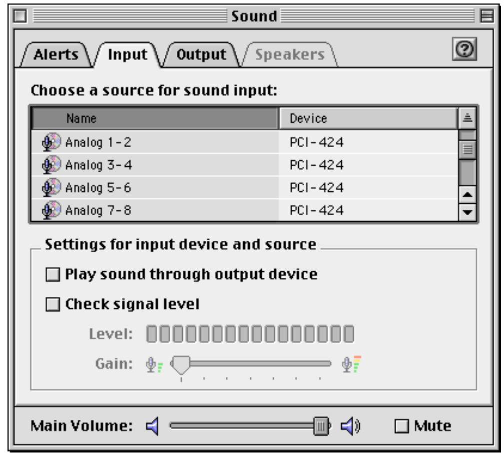
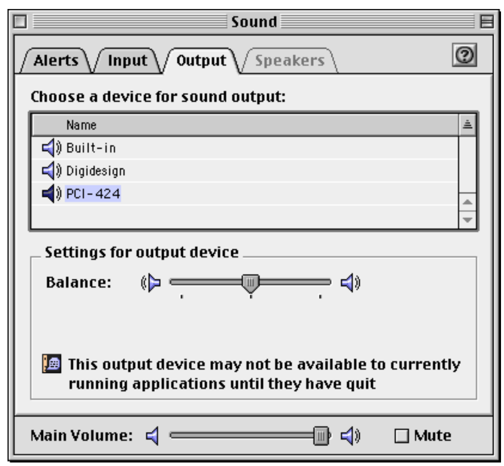


Figure 10-4: Choosing the desired input for Sound Manager. Any input pairs that are currently checked in the MOTU PCI Audio Console will appear in this list.

5 Click the *Output* tab and choose the PCI-424. To learn how to specify the output pair, see the next section, "Changing the output".



6 Make sure that the *Main Volume* setting is not muted. If your audio software has its own volume control for Sound Manager output, raise this volume slider to full volume. You can attenuate it as needed in your audio software.

That's it. You can now run any Sound Manager compatible audio software, and play back audio through the PCI-424 outputs you've chosen in the MOTU PCI Audio Console *Monitor Outputs* menu.

CHANGING THE OUTPUT

To change the PCI-424 output being used by Sound Manager, run the MOTU PCI Audio Console, and choose the desired output pair from the *Monitor Outputs* menu as shown in Figure 10-2 on page 98.

SOUND MANAGER AND 2408MK3 STAND-ALONE OPERATION

As discussed earlier in chapter 5, "2408mk3 Front Panel Operation" (page 57), the 2408mk3 has two modes: PCI mode and stand-alone mode. It goes into PCI mode as soon as any 2408mk3-related software is launched on the Macintosh. It remains in PCI mode until all 2408mk3-related software quits. Only then can it operate in stand-alone mode.

If you've installed the Sound Manager driver as described in this chapter, your 2408mk3 hardware will go into PCI mode as soon as you start up your Macintosh (since the MOTU PCI Sound Manager driver loads at startup, like the rest of the System Extensions). The 2408mk3 hardware will stay in PCI mode until you shut down your computer.

Therefore, if you would like to use the 2408mk3 in stand-alone mode, either shut down your computer, or simply restart it with the MOTU PCI Sound Manager Driver extension disabled. See the next section for how to do so.

MONITORING LIVE INPUTS

If you would like to listen to live inputs as you are recording them into your Sound Manager application, you need to use the PCI-424's CueMix feature. For details, see "Using CueMix with Sound Manager" on page 109 and chapter 13, "CueMix Console" (page 111).

DEACTIVATING THE DRIVER

You can deactivate the MOTU PCI Sound Manager driver at any time by choosing the Extensions Manager Control Panel and unchecking it in the list.

AUDIOTAP AND DIGITAL PERFORMER

Digital Performer includes a MOTU Audio System plug-in called AudioTap that allows you to route any Sound Manager audio into Digital Performer's mixing environment. From there, you can route it to your PCI-424 interface via any of Digital Performer's extensive audio routing features. For details, consult your Digital Performer documentation.

CHAPTER 11 Expanding Your PCI-424 System

OVERVIEW

Up to four MOTU PCI audio interfaces can be connected to a single PCI-424 card for up to 96 simultaneously active inputs and outputs. For example, if you connect four 2408mk3 interfaces, you get a total of 232 and 248 physical input and output connections (respectively) and up to 96 input and output channels. You can freely "mix and match" any MOTU PCI audio interfaces together on a single PCI-424 card, including the 24I/O, HD192, 2408, 2408mkII, 1296, 1224, 24i and 308 models.

The PCI-424 card, with its CueMix DSP engine, ties all connected audio interfaces together, allowing them to act as a massive audio routing matrix for mixing, merging, splitting and routing to/from all connected inputs and outputs.

MOTU PCI audio interfaces are sold separately as expanders for any MOTU core system. See your MOTU dealer for details.

CONNECTING EXPANSION INTERFACES

Connect expansion interfaces to the three available Audio Wire sockets on your core system's PCI-424 card as shown below in Figure 11-1.

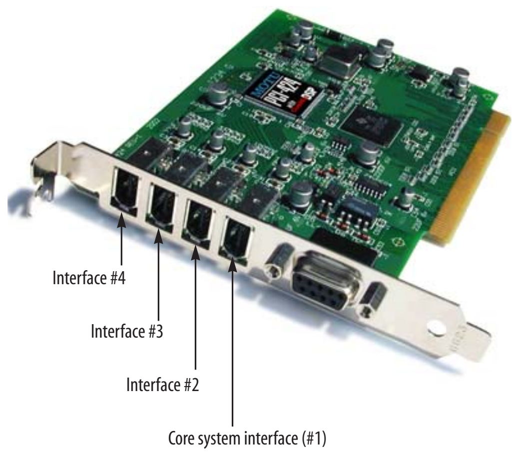


Figure 11-1: You can connect up to four MOTU PCI audio interfaces to a single PCI-424 card.

MULTIPLE INTERFACES IN THE CONSOLE

The MOTU PCI Audio Console window displays the settings for one PCI-424 interface at a time. To choose which I/O you are looking at, choose it from the *Configure Interface* menu as shown below in Figure 11-2.

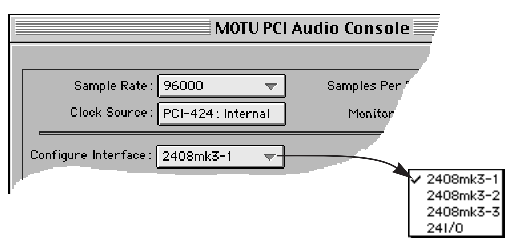
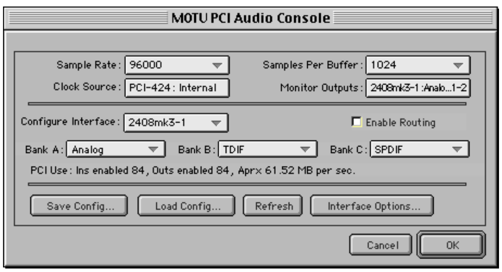


Figure 11-2: Choosing which audio interface you are working with in the MOTU PCI Audio Console window.

Enabling banks, or individual ins and outs

Bank enabling, or enabling individual inputs and outputs, works the same for each interface in the MOTU PCI Audio Console window. Choose an interface from the Configure Interface menu (Figure 11-2) and then configure the banks (or individual outputs) as usual. See “Interface settings” on page 70 and “Enable routing” on page 71 for details.



Working with input and output menus

With multiple interfaces connected, input and output menus show all available ports on all interfaces, as demonstrated below in Figure 11-3. If interfaces are configured to share the same type of I/O format (analog, for example), their outputs are labeled with the bank letter (A, B or C) and interface name, along with numbered suffixes (e.g. 2408mk3-1, 2408mk3-2, etc.) according to the order in which they are connected to the PCI-424 card as shown in Figure 11-1 on page 101.

Notice that in the example below, the second 2408mk3 interface is configured for three ADAT banks, but since no other interface is configured for the ADAT format, the interface name is not included in the name of the input or output.

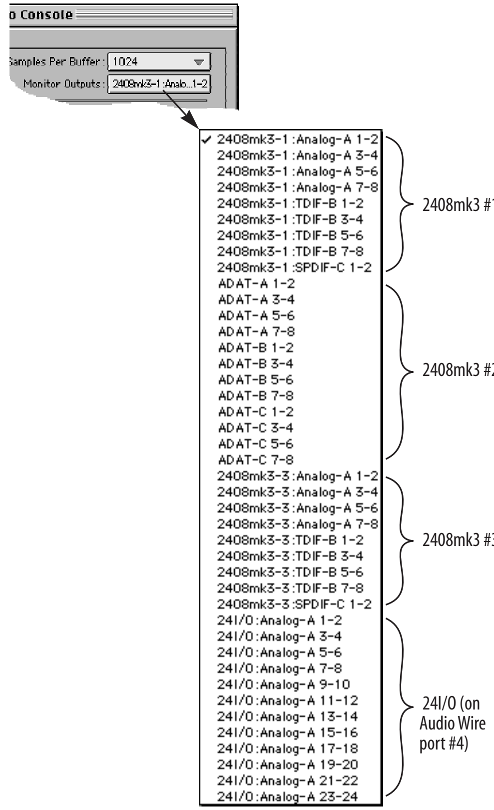


Figure 11-3: Input and output menus show the ports of all connected interfaces. In this example, there are three 2408mk3 interfaces and one 24I/O interface.

CUEMIX DSP ROUTING AMONG INTERFACES

The PCI-424 card, with its CueMix DSP engine, ties all connected audio interfaces together, allowing them to act as a massive audio routing matrix for mixing, merging, splitting and routing to/from all connected inputs and outputs. You can route any input to any output, create multiple mix busses and mix inputs to outputs across all interfaces. For details, see chapter 13, “CueMix Console” (page 111).

SYNCHRONIZING MULTIPLE INTERFACES

The entire PCI-424 system gets its clock from whatever you choose from the *Clock Source* menu in the MOTU PCI Audio Console window. Synchronization across all connected interfaces is sample-accurate. All available sync sources on all connected interfaces are displayed in the menu as demonstrated below in Figure 11-4.

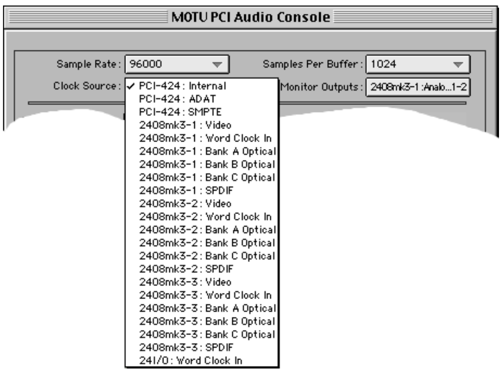


Figure 11-4: The entire PCI-424 system can be clocked from any sync source on any connected interface. After you choose a source from this menu, the entire system, including all connected interfaces, synchronizes to it with sample-accurate precision. This example shows three 2408mk3 interfaces and one 24I/O interface.

Word clock connections are not necessary

Each MOTU PCI audio interface in the system gets its clock from the Audio Wire cable connection to the PCI-424 card (unless it is the master clock itself). There is no need to make word clock connections between interfaces.

SAMPLE RATE ISSUES

While the PCI-424 card can operate at sample rates as high as 192 kHz, not all MOTU PCI audio interfaces support the higher rates. When choosing high sample rates (above 48 kHz) interfaces that do not support the rate will be temporarily taken off line by the system. For complete details, see "Clock Source" on page 66.

PCI BANDWIDTH ISSUES

Attempting to run 96 channels of 96 kHz audio input and output (that's 192 channels total) places extreme demands on your computer's PCI bus. Today's fastest computers can just barely handle this much data on their PCI bus. Slower computers will lower the number of banks you can use at one time. See "Disabling banks to conserve resources" on page 70 and "PCI use" on page 71 to learn more about managing your system's PCI resources.

CHAPTER 12 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 a reverb 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 a 2408mk3 input, passes through the 2408mk3 hardware into the computer, through your host audio software, and then back out to a 2408mk3 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 PCI-424's CueMix DSP feature to patch the input directly to your monitor outs via the PCI-424 audio hardware. This is just like bussing inputs to outputs in a digital mixer. For details, see "CueMix DSP hardware monitoring" on page 108.

If you *do* need to process a live input with plug-ins, or if you are playing virtual instruments live through your MOTU PCI 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.

MONITORING LIVE INPUT

There are two ways to monitor live audio input with a MOTU PCI-424 system: 1) through the computer or 2) via CueMix™ DSP hardware monitoring. Figure 12-1 on page 106 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, "Samples Per Buffer" for details about how to reduce — and possibly eliminate — the audible monitoring delay that the computer introduces.

Figure 12-2 shows how to use CueMix™ DSP hardware-based monitoring, which lets you hear what you are recording with no monitoring delay and no computer-based effects processing. (You can add effects later, after you've recorded the live input as a disk track.) See "CueMix DSP hardware monitoring" later in this chapter for details on how to use CueMix DSP with your audio software, or with the included CueMix Console software.

If the material you are recording is suitable, there is a third way to monitor live input: use both methods (Figure 12-1 and Figure 12-2) at the same time. For example, you could route vocals to both the computer (for a bit of reverb) and mix that processed signal on the main outs with dry vocals from CueMix DSP.

THE SAMPLES PER BUFFER SETTING

As shown in Figure 12-3, the *Samples Per Buffer* setting determines the size of the buffers used by the PCI-424 drivers to transfer audio to and from the PCI-424 hardware. This setting can be used to reduce latency.

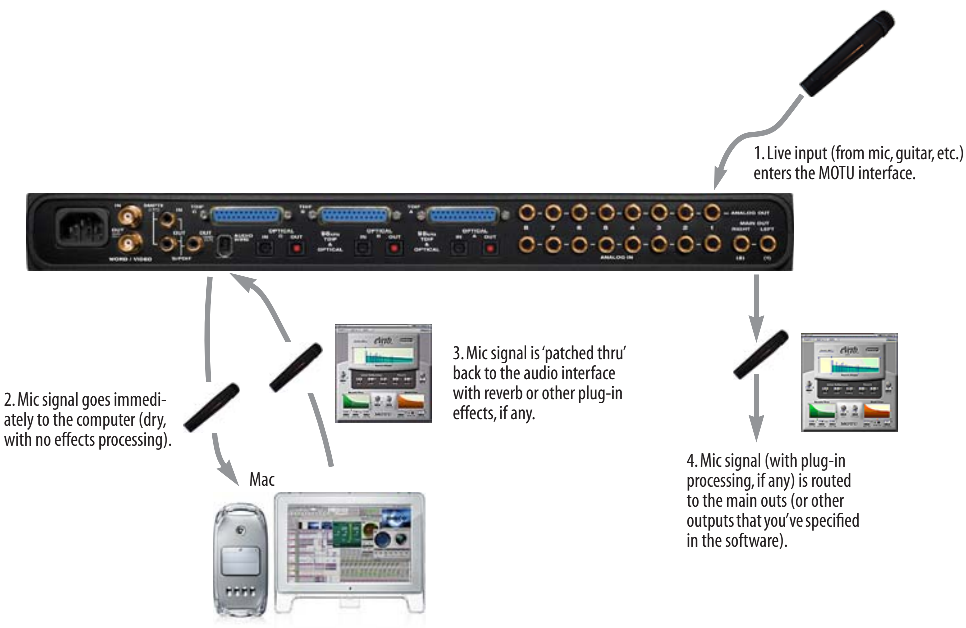


Figure 12-1: There are two ways to monitor live audio inputs with a PCI-424 system: 1) through the computer or 2) via CueMix™ DSP hardware monitoring. This diagram shows method 1 (through the computer). When using this method, use the PCI-424's 'Samples Per Buffer' setting to reduce the slight delay you hear when monitoring the live input, but don't lower it too much, or your computer will act sluggish.

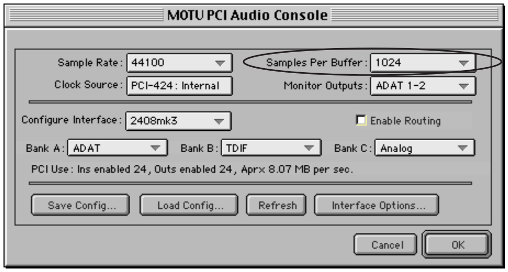


Figure 12-3: Lowering the 'Samples Per Buffer' setting in the MOTU PCI Audio Console Window reduces patch thru latency. But doing so increases the processing load on your computer, so keep an eye on the Performance Monitor window inAudioDesk (or similar feature in your host audio software).

For details on how to access the MOTU PCI Audio Console window show above in Figure 12-3, see "Accessing the PCI Audio Console window" on page 65.

The Samples Per 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 AudioDesk, Digital Performer or other audio software

Lower latency versus higher CPU overhead

The *Samples Per 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 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.

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,

Figure 12-2: This diagram shows the signal flow when using CueMix™ DSP no-latency monitoring. Notice that this method does not allow you to process the live input with plug-ins in your audio software while it is being monitored. You can, however, add effects later — after recording the live input as a disk track. CueMix™ DSP lets you hear what you are recording with no delay and no computer-based effects.

choose a higher Samples Per Buffer setting. Depending on your computer's CPU speed, you might find that settings in the middle work best (256 to 2048).

Transport responsiveness

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.

Effects processing and automated mixing

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

CUEMIX DSP HARDWARE MONITORING

The PCI-424 card has a more direct method of patching audio through the system. This method is called *CueMix DSP*. When enabled, CueMix activates hardware patch-thru in the PCI-424 card itself. CueMix DSP has two important benefits:

- First, it completely eliminates the patch thru delay (reducing it to a small number of samples — about the same amount as one of today's digital mixers).
- Secondly, CueMix DSP imposes no strain on the computer.

The trade-off, however, is that CueMix DSP bypasses your host audio software. Instead, live audio inputs are patched directly through to outputs in the PCI-424 card 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 DSP 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 DSP. Instead, reduce latency with the *Samples Per Buffer* setting (as explained earlier in this chapter).

TWO METHODS FOR CONTROLLING CUEMIX DSP

If you are using Digital Performer, AudioDesk or ASIO-compatible audio software, there are two ways to control CueMix DSP:

- From AudioDesk, Digital Performer, or your ASIO-compatible host software
- From CueMix Console

You can even use both methods simultaneously.

Controlling CueMix DSP from your audio software

Follow the directions below in the section that applies to you.

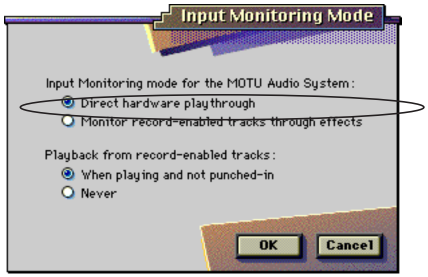
Controlling CueMix from within AudioDesk or Digital Performer

To turn on CueMix DSP in AudioDesk and Digital Performer:

1 From the Basics menu, choose *MOTU Audio System options>Input Monitoring Mode*.

2 Choose the *Direct hardware playthrough* option, as shown below in Figure 12-4.

3 From the Windows menu, choose *Audio Monitor*, and enable Audio Patch Thru (the button with the headphone icon on it).



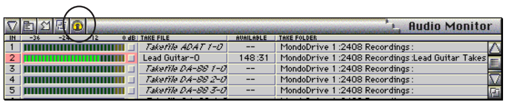


Figure 12-4: Enabling CueMix DSP in AudioDesk or Digital Performer.

Once enabled, CueMix DSP monitoring is tied with Digital Performer or AudioDesk's *Audio Patch Thru* feature: when you record-enable a track, the track's input is routed directly to its output (via CueMix DSP in the PCI-424 hardware). This connection is made "under the hood", which means that you won't see it in CueMix Console. However, CueMix DSP connections made inside Digital Performer or AudioDesk will dovetail with any other mixes you've set up in CueMix Console. For example, if a record-enabled track is assigned to an output pair that is already being used in CueMix Console for an entirely separate mix bus, the track's patched thru audio will simply be merged with the CueMix Console mix bus output.

☛ Note to PCI-324 users who have upgraded to a PCI-424 system: notice that the *Auto Cuemix Update* check box has been removed as a result of the PCI-424's more flexible and powerful CueMix DSP features. *Auto CueMix Update* is no longer needed because you enjoy the benefits of CueMix DSP patch thru, *plus* separate, independent mixing under CueMix Console, thanks to the much more powerful CueMix DSP engine.

Using CueMix with Sound Manager

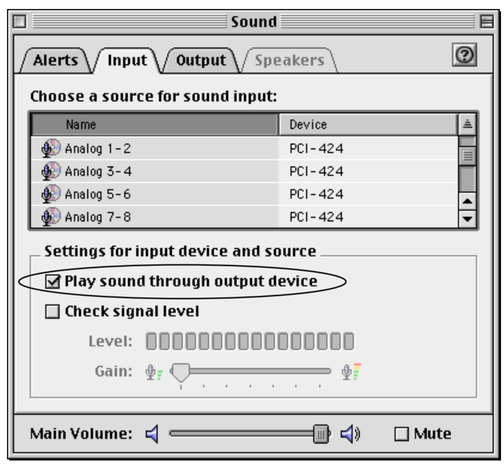
To enable CueMix DSP for a host application that is using the MOTU PCI Sound Manager driver:

1 Make all of the Sound Manager preparations described in chapter 10, "Sound Manager" (page 97).

2 Choose the desired output for Sound Manager in the MOTU PCI Audio Console window, as explained in "Monitor Outputs" on page 69.

3 Open the Sound Control Panel, click the *Input* tab, and check the *Play sound through output device* option. Alternately, you can use the CueMix Console (described in chapter 13, "CueMix Console" (page 111)) to manually patch a live input to an output.

4 To control the overall level of the CueMix input, Use the CueMix Console.



Controlling CueMix DSP from within Cubase
To turn on CueMix in Cubase VST, enable the *ASIO Direct Monitor* check box in the Monitoring section of the Audio System Setup window (Figure 9-3 on page 90).

Once enabled, CueMix DSP monitoring is controlled with Cubase's input monitoring features: when you enable monitoring for a track, the track's input is routed directly to its output fader (via CueMix DSP in the PCI-424 hardware). This connection is made "under the hood", which means that you won't see it in CueMix Console. However, CueMix DSP monitoring connections made inside Cubase will dovetail with any other mixes you've set up in CueMix Console. For example, if monitoring is enabled for track assigned to an output pair that is already being used in CueMix Console for an entirely separate mix bus, the Cubase-enabled audio will simply be merged with the CueMix Console mix bus output.

☛ Note to PCI-324 users who have upgraded to a PCI-424 system: CueMix DSP monitoring in Cubase no longer affects the settings in CueMix Console. Instead, you now enjoy the benefits of CueMix DSP monitoring in Cubase, *plus* separate, independent mixing under CueMix Console, thanks to the much more powerful CueMix DSP engine.

Other ASIO-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 Console
If you are not using ASIO compatible software, or if your host audio software does not support ASIO's direct hardware monitoring features, you can set up your monitor mix in CueMix Console.

CueMix Console allows you to set up monitor mixes, or any other desired routing configurations, that are independent of your host audio software. See chapter 13, "CueMix Console" (page 111).

CHAPTER 13 CueMix Console

OVERVIEW

CueMix Console provides access to the flexible on-board mixing features of the PCI-424 system. CueMix lets you route any combination of inputs to any stereo output pair. These mixes can be set up entirely independently of your host audio software. CueMix allows you to set up as many separate mixes as there are stereo outputs in your system. You can also save and load mix configurations.

CueMix Console can be used independently of host audio software, or together with it. CueMix mixing dovetails with the direct monitoring (hardware patch thru) features of your host audio software, allowing you to seemlessly mix in both environments.

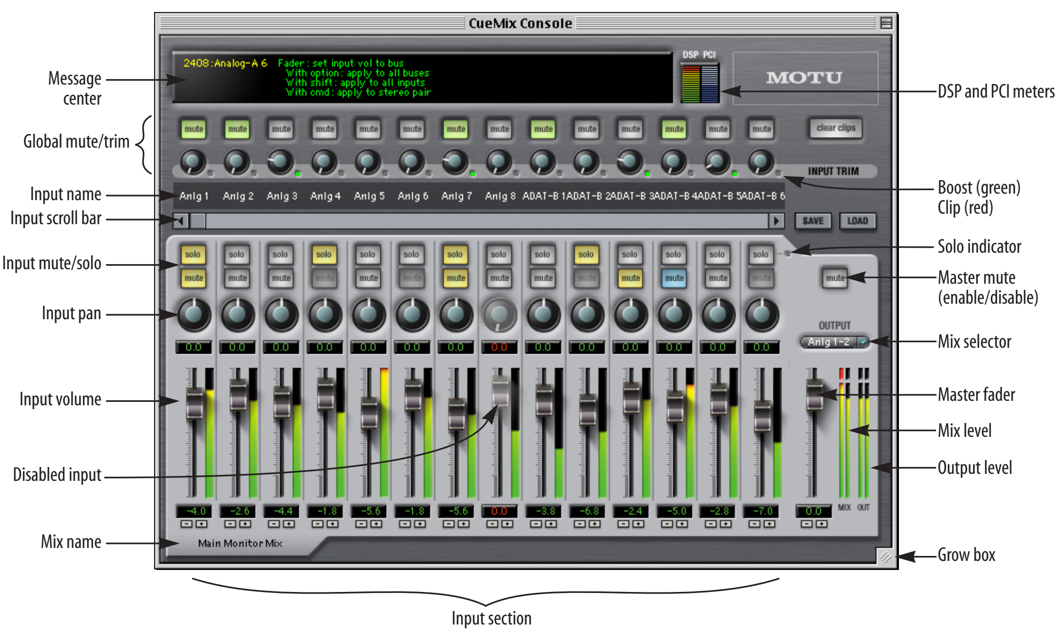


Figure 13-1: CueMix Console is a virtual mixer that gives you control over the PCI-424 card's on-board mixing features.

ADVANTAGES OF CUEMIX MONITORING

CueMix Console provides several major advantages over monitoring live inputs through your host audio software:

- CueMix has no buffer latency. Thanks to the PCI-424 card's DSP chip, CueMix provides the same throughput performance as a digital mixer.
- CueMix imposes absolutely no processor drain on the computer's CPU.
- CueMix routing can be maintained independently of individual software applications or projects.

CueMix Console does not provide effects processing. For information about using your audio software's native plug-ins together with CueMix, see chapter 12, "Reducing Monitoring Latency" (page 105).

CUEMIX CONSOLE INSTALLATION

CueMix Console is installed with the rest of your PCI-424 software on the top level of your startup hard drive.

CUEMIX CONSOLE BASIC OPERATION

The CueMix console is simple to operate, once you understand these basic concepts.

One mix per stereo output pair

CueMix provides one mix for every physical stereo output pair in your system. For example, if you have a core 2408mk3 system with all three banks enabled for 8-channel I/O (using any available format), your system has 24 outputs. Therefore, CueMix Console provides you with 12 mixes (12 stereo output pairs). Each mix is identified by its stereo outputs (e.g. *Analog 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 Console 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 Console displays one mix at a time. To select which mix you are viewing, choose it from the Mix Selector menu above the master fader, as shown in Figure 13-1. The mix name appears in the tab at the bottom of the window. Double-click the name to change it.

Each mix is completely independent

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

Widening the CueMix Console window

To view more input faders at once, drag the grow box (Figure 13-1) to the right.

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

These elements are visually grouped together in the lightly shaded area in the lower half of the CueMix Console window.

Choosing a mix
To view mix, choose it from the Mix Selector menu above the master fader, as shown in Figure 13-1. The mix name appears in the tab at the bottom of the window.

Naming a mix
Choose the mix from the Mix Selector menu, and then double-click the name tab below the channel strip section (Figure 13-1) to type in a new name.

Master mute
The master mute button (Figure 13-1) temporarily disables (silences) the mix.

Master fader
The master fader (Figure 13-1) 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 and Output level meters
There are two sets of level meters to the right of the master fader: MIX and OUT. The MIX meters show you the overall mix output. The OUT level meters show you the level of the physical outputs for the mix, which may include audio from your host audio software. Use the CLEAR CLIPs button to clear the clip indicators on these meters.

Input section
The channel strips to the left of the master fader represent each input in your PCI-424 system. Use the input scroll bar to view additional inputs. All inputs currently available on all connected interfaces will be available in this scrollable section of the window.

Input mute/solo
To add an input to a mix, or remove it, click its MUTE button. To solo it, use its SOLO button. To toggle these buttons for a stereo pair, hold down the command key while clicking either channel.

The Solo indicator LED (Figure 13-1) lights up when any input is soloed (including inputs that may currently be scrolled off-screen).

☛ The Global Mute buttons above the channel strips (just below the message center) enable or disable the input globally. For details, see “Global mute and trim” on page 114.

Input volume and pan
Use the input fader and pan knob (Figure 13-1) to adjust these settings for the input in the mix. Again, all settings within the gray-shaded channel strip area belong to the mix currently being viewed. Note that an input can have different settings in different mixes.

To adjust the volume or panning for a stereo input pair, hold down the command key while dragging the fader or knob for either the left or right input.

SHORTCUTS
Hold down the following modifier keys as shortcuts:

Shortcut	Result
Shift key	Applies your action to all inputs in the mix.
Command key	Applies your action to the stereo input pair
Option key	Applies your action to all busses
Double-click	Returns the control to its default value (pan center, unity gain, etc.)

COPYING & PASTING (DUPLICATING) ENTIRE MIXES
To copy and paste the settings from one mix to another:

1 Select the source mix from the Mix Selector menu (Figure 13-1) and choose Copy from the file menu (or press command-C).

2 Choose the destination mix from the Mix Selector and choose Paste from the file menu (or press command-V).

MESSAGE CENTER

The Message Center displays fly-over help for items in the CueMix Console window, as well as information about the current PCI bus usage in your computer. It also displays messages regarding the overall operation of the PCI-424 system.

GLOBAL MUTE AND TRIM

The MUTE and trim controls at the very top of the CueMix Console window (Figure 13-1) affect each input globally, across all mixes and for audio being routed to the computer. For example, if you globally mute an input, it is temporarily muted in all mixes, and your host software will not receive any audio from that input, either.

Up to 12 dB of boost with the global trim

The global trim knob for each input (Figure 13-1) provides up to 12 dB of boost. This setting is applied globally for the input. For zero boost, turn the knob all the way down until the green boost LED becomes dark.

Input clip indicator

The LED to the right of each input's global trim knob (Figure 13-1) turns red when clipping occurs on the input and stays yellow for a few seconds.

Clear Clips

This button clears the "over" LEDs on the front panel of an HD192 or 1296 interface. It also clears the (temporary) clip LEDs in CueMix Console.

SAVING AND LOADING CUEMIX CONFIGURATIONS

The Save and Load buttons above the master fader allow you to save CueMix console configurations. This can be particularly convenient if you have multiple MOTU audio interfaces connected to your system and have several operating scenarios in regards to managing live input mixing. Click the Save button to save the current configuration; click the Load button to open an existing configuration that you have previously saved on disk.

MANAGING CUEMIX DSP RESOURCES

CueMix DSP's resources are not unlimited: every time you enable an input, it requires a small amount of the PCI-424 card's DSP bandwidth. At the time of this writing, CueMix DSP is still being tested and optimized. It appears to be able to support at least 96 total inputs (across any/all mixes), and perhaps more. For example, you could enable 12 mixes and assign up to 8 inputs each, or 6 mixes with 16 inputs each, or any combination that totals 96 faders.

Grayed out faders

If you exceed CueMix DSP's bandwidth limit, one input will become temporarily disabled for each input you enable over the limit. CueMix DSP inputs that are enabled from within your MOTU or ASIO host software (see "Controlling CueMix DSP from your audio software" on page 108) are given highest priority, followed by CueMix Console inputs, which gray out as shown in Figure 13-1 when they are temporarily disabled. In addition, a disabled fader displays its level readout in red.

DSP meter

The DSP meter in CueMix Console gives you a rough idea of how much CueMix DSP bandwidth is used up and how much is still available. If you are using a lot of CueMix DSP inputs, keep your eye on this meter for an idea of when you'll run out.

PCI Meter

The PCI meter gives you a rough idea of how much PCI bandwidth is used up and how much is still available. The PCI-424 card often shares this bandwidth with other PCI cards, or other components in your computer. The light blue portion indicates overall PCI use (from all devices using the bus), while the dark blue portion indicates how much bandwidth the PCI-424 card is using. If this meter gets close to peaking, disable other PCI devices, or disable banks and/or audio channels in your PCI-424 system using the MOTU PCI Audio Console.

CUEMIX CONSOLE EXAMPLES

Figure 13-2 below shows some examples of how you can use CueMix DSP:

■ All of these live inputs can be bussed to a pair of powered monitors connected directly to a PCI-424 interface. For example, if the monitors were connected to the main analog outs of a 2408mk3 interface, you could name it "Main Outs" and enable the inputs any/all of the sources shown here.

■ 8 channels of GigaStudio running on a PC laptop are being fed into the PCI-424 system via an optical connection between the 828 and the 2408mk3. These Giga inputs are being monitored live on the studio monitors with only a few samples of latency from start to finish, thanks to the all-digital signal path.

■ The Roland XV-5080 and Kurzweil K2500 represent how you can feed the live input from synthesizers, drum machines and other MIDI instruments — or any live inputs — into the system and monitor them live via connected studio monitors.

■ Connect dedicated, premium mic preamps, such as the Avalon preamp show, and route live mics and guitars directly into the PCI-424 system and bus them to your monitors.

■ The Lexicon MPX-1 represents a hardware effects loop. Dedicate a mix to it in CueMix Console and then route any live input to it as needed with no latency. You can monitor the result live via your studio monitors (main output mix) and even record (print) the effects in your software.

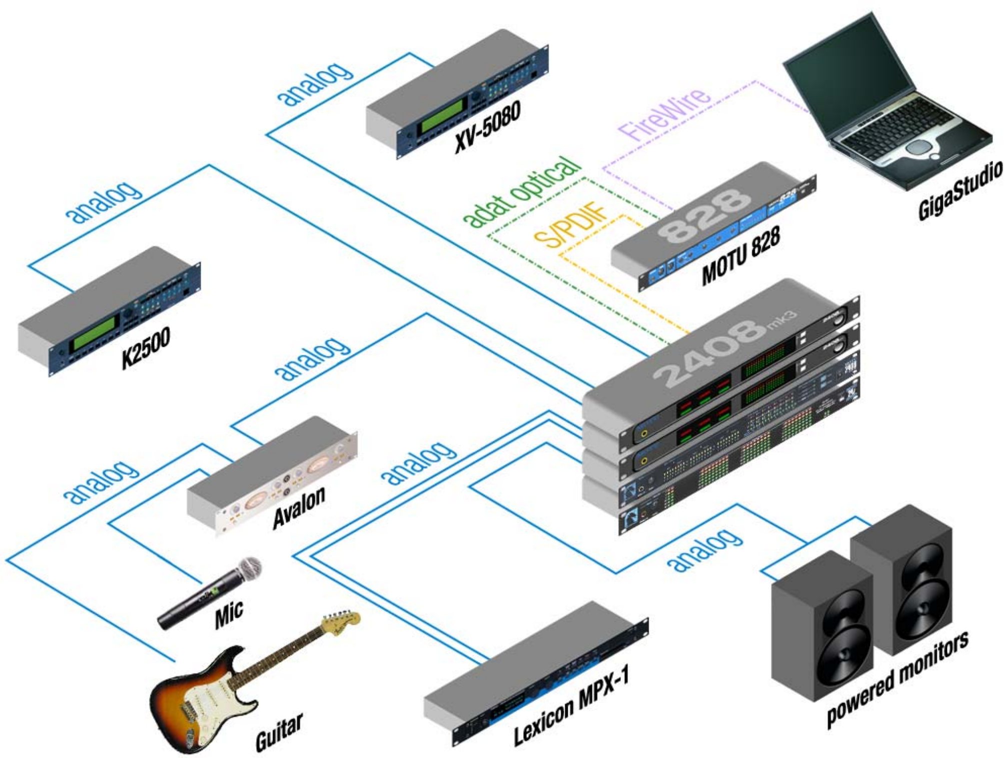


Figure 13-2: An example setup of a system that takes full advantage of CueMix DSP.

CHAPTER 14 MOTU SMPTE Console

OVERVIEW

The PCI-424 system can resolve to SMPTE time code, without a dedicated synchronizer. It can also serve as a SMPTE time code generator. If you have a 2408mk3 interface connected, you can also resolve the entire system to video, without a dedicated synchronizer.

The MOTU SMPTE Console software provides a complete set of tools to resolve to video and SMPTE, and to generate SMPTE for striping, regenerating or slaving other devices to the computer.

The PCI–424 card provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy. Supported video formats include NTSC, PAL/SECAM or blackburst.

The pair of RCA S/PDIF jacks on the 2408mk3 interface rear panel can be switched via the SMPTE Console software to become a dedicated SMPTE time code (LTC) audio input and output. However, because the 2408mk3's sync features are driven by the PCI-424 on-board DSP, any analog input on any available interface can be chosen for SMPTE input, and any active channel, digital or analog, can be chosen as a SMPTE time code output.

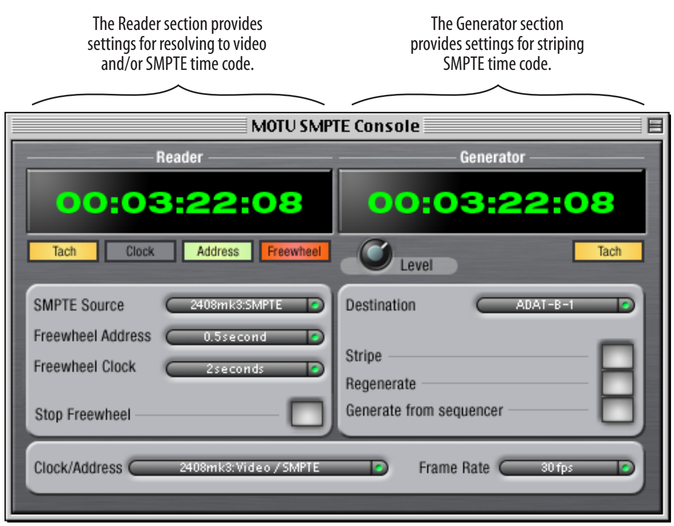


Figure 14-1: SMPTE Console gives you access to your PCI-424 system's on-board video and SMPTE time code synchronization features.

CLOCK/ADDRESS

The Clock/Address menu provides the same global Clock Source setting as in the MOTU PCI Audio Console ("Clock Source" on page 66), but it includes additional information: each setting shows both the clock and the address (time code or sample location), separated by a forward slash (/). For example, the word clock setting (*2408mk3:Word Clock In / Internal*) shows the clock source (*Word Clock In*) followed by the address (*Internal*). Notice that only the SMPTE and video settings support SMPTE time code as the reference for address. The digital clock sources do not support the PCI-424 system's on-board SMPTE time code sync.

Resolving to SMPTE time code

To resolve the PCI-424 system to SMPTE time code, choose the *PCI-424: 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. Be sure to specify the input that is receiving the SMPTE time code by choosing it from *SMPTE source* menu. For further details on the hardware connections involved, see "Syncing to SMPTE time code" on page 41.

Resolving to video

To resolve the PCI-424 system directly to video, you need a 2408mk3 interface. Choose the *PCI-424: Video / SMPTE* setting in the *Clock/ Address* menu. The system will resolve to the video signal received on the Video In BNC jack on the 2408mk3 interface for clock (time base), and it will reference SMPTE time code (on the analog input specified in the *SMPTE Source* menu) for address. For further details on the hardware connections involved, see "Syncing to video" on page 42.

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 PCI-424 system can auto-detect and switch to the incoming frame rate, except that it cannot distinguish between 30 fps and 29.97 fps time code. So if you are working with either of these rates, make sure you choose the correct rate from this menu. The PCI-424 driver updates the frame rate setting in Digital Performer and AudioDesk for you. For ASIO applications, however, you need to make sure that their frame rate it set properly.

READER SECTION

The Reader section (on the left-hand side of the window in Figure 14-1) provides settings for synchronizing the PCI-424 system to video and/or 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 PCI-424 system has successfully achieved lockup to SMPTE time code and SMPTE frame locations are being read.

Clock

The Clock light glows continuously when the PCI-424 system has successfully achieved lockup to an external time base, either SMPTE time code or video.

Address

The Address light glows continuously when the PCI-424 system has successfully achieved lockup to SMPTE time code.

Freewheel

The Freewheel light illuminates when the PCI-424 system 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 will be receiving the SMPTE time code from this menu. The *2408mk3:SMPTE* setting refers to the RCA input jack on the rear of the 2408mk3 interface that doubles as both a dedicated (analog) SMPTE time code input and a SPDIF input. As soon as you choose the *2408mk3:SMPTE* setting, the jack automatically switches internally to an analog time code circuit.

Freewheel Address

Freewheeling occurs when there is a glitch or drop-out in the incoming time code for some reason. The PCI-424 system 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 PCI-424 system to freewheel before it gives up and stops altogether.

The PCI-424 system 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 PCI-424 system 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 or video signal for some reason. The PCI-424 can freewheel past the drop-out and then resume lockup again as soon as it receives a stable, readable clock signal.

The PCI-424 system 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 PCI-424 system 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 14-1) provides setting for generating SMPTE time code.

Tach light

The Tach light blinks once per second when the PCI-424 system is generating SMPTE time code.

Level

Adjusts the volume of the SMPTE time code output.

Destination

Lets you choose any available analog or digital output in your PCI-424 system. The choices you see in this menu correspond to the banks (and individual outputs) you've enabled in the MOTU PCI Audio Console. See "Interface settings" on page 70.

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.

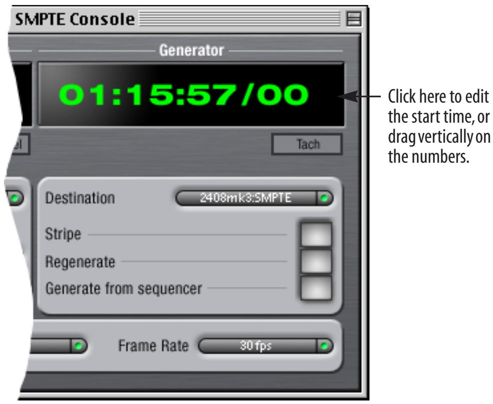


Figure 14-2: Setting the time code start time.

Regenerate

This option, when enabled, causes the generator to generate time code whenever the PCI-424 system is receiving either SMPTE time code or ADAT Sync (via its ADAT Sync In port).

Generate from sequencer

This option, when enabled, causes the generator to generate time code whenever you are running AudioDesk or Digital Performer. Time code begins at the time specified by the AudioDesk or Digital Performer main transport.

CHAPTER 15 Troubleshooting

Using Pro Tools, Sound Manager and -50 error
When using Sound Manager, Pro Tools software will only allow audio input via the Macintosh's Built-in hardware. Therefore, you cannot use the PCI-424 system as the input device to Pro Tools software. If the PCI-424 Driver is selected as the input device in the Sound Control Panel, Pro Tools will return a -50 error and not launch. You can, however, select Built-in as the input device and the PCI-424 Sound Manager Driver as the output device in the Sound Control Panel. After doing so, you can run Pro Tools and monitor your output through the PCI-424 system.

My host software doesn't receive any audio from an input; other inputs are working fine, but this one input isn't.
Open CueMix Console and make sure that the input isn't globally muted (the mute button at the very top of the channel strip, below the message center).

Sample accurate sync in AudioDesk and Digital Performer
When you first use sample accurate sync, be sure to go to the Receive Sync dialog in Digital Performer or AudioDesk and switch from "MTC" to "Sample-accurate."

Cubase - Inputs and outputs are enabled in the MOTU PCI Audio Console, but still not visible in Cubase
This would indicate that you do not have these inputs enabled in Cubase. Select Inputs from the Audio menu. Make sure that the inputs are enabled.

Monitoring input sources via the PCI-424 Sound Manager driver
With the PCI-424 system, monitoring input sources via the Sound Manager Driver takes place at the hardware level. Be sure the PCI-424 Sound Manager Driver is chosen in the Sound Control Panel for input and output. If you enable Monitoring in a Sound Manager application like PEAK 2.0, you must also set the level for the output pair using the PCI-424 CueMix Console.

Can't authenticate AudioDesk
When installing software off the CD, the OK button does not become active until you have entered in your name and a valid keycode. Your name must contain at least 3 characters, and you must enter the keycode exactly as it appears in your users manual.

MOTU PCI Audio Console - Error: The PCI-424 driver can not be accessed at this time because it is currently in use
This error is caused when another application is using the PCI-424 driver at the while you attempt to configure your MOTU Audio interface via the MOTU PCI Audio Console. While the PCI-424 driver is a multi-client driver and several applications can access the driver at once, you cannot configure the driver while it is in use by another application. If you have the PCI-424 Sound Manager driver enabled for Sound In and Sound Out, and there no applications running besides Digital Performer or Audio Desk, it is likely that an extension may be claiming the Sound Manager driver when the computer boots up. The solution: Disable the extension or choose Built-In for both Sound In and Sound Out in the Sound Control Panel.

ADAT-compatible or Tascam tape decks - converting 48kHz tracks to 44.1kHz

If the audio on your ADAT or Tascam tape was recorded at 48kHz, then you must transfer that audio into your host software at 48kHz. If you plan to digitally mix this data and burn an audio CD, you must convert your audio to 44.1kHz with your host software. Once you sample rate convert your audio to 44.1kHz, you will have to set the sample rate in the console back to 44100 so you can play the file. You always want your digital clocks and sample rates to match when dealing with any kind of digital audio transfer or synchronizing.

ADAT-compatible or Tascam tape decks - device order in MIDI Machine Control window

When powered on, most modular digital multi tracks 'wake up' configured to record from their analog inputs by default. You won't be able to record from the PCI-424 system to a MDM until it is switched to digital input. Tip: configure this in ClockWorks or AudioDesk if you want your decks to come up in the right mode when powered on.

The order of devices in the DTP panel of the MIDI Machine Control window is not controlled by the order of connections to the PCI-424 system. This can be confusing unless you make them match when connecting the system.With Tascam units, make sure to assign their hardware IDs in the order that they are connected to 2408mk3 digital I/O banks. With ADATs make sure to chain their sync connections in the order that they are connected to 2408mk3 banks.

No input from an ADAT or Tascam tape deck

If you are having trouble recording on your ADAT or Tascam tape deck from the 2408mk3, check the Digital input setting. After power cycling, tape decks often come up configured to record from their analog inputs. You won't be able to record from the 2408mk3 to a tape deck until it is switched to digital input. Tip: configure this in ClockWorks or AudioDesk if you want your decks to come up in the right mode when power cycled.

ADATs and the DTP - lock up time allowances

ADATs can take a while to sync to the DTP. For example, when recording from a 2408mk3 to the ADATs, they may appear to chase and lock, but the record button continues to flash. Recorded data on the tape won't be sample accurate until the record light stops flashing. Solution: add more pre-roll time.

Alesis blackface (classic) ADAT optical sync

If you are using a Classic (black-faced) ADAT as a sync master in your PCI-424 system, and you want to record sample accurately into it, you must configure it to use its internal clock, instead of slaving to the optical input. To do this, hold down the SET LOCATE button and press the DIGITAL IN button. This will toggle the ADAT between external sync (shown as "diG") and internal sync ("int"). Note that the ADAT will return to the "diG" setting each time it is powered off.

Alesis ADAT track offset

If you find that sample accurate transfers from an ADAT are displaced by a small amount, try to reset the ADAT to its factory default state. This is accomplished by powering on the ADAT while holding down the RECORD and PLAY keys simultaneously. The ADAT LX uses a 'soft' power key so LX users will have to unplug and plug the ADAT LX while holding down the RECORD and PLAY keys.

Tascam TDIF and word clock

If any Tascam TDIF units are connected, they must be word locked to the PCI-424 system. The Tascam TDIF protocol does not contain word clock, so you must slave the Tascam to the 2408mk3's word clock output. For detailed instructions on how to accomplish this, see "Syncing Tascam 'TDIF' devices" on page 47, "Syncing to a DA-88/98 time code track" on page 45, or "Sample-accurate

Tascam sync" on page 40. Another alternative is to slave everything to a Digital Timepiece. The Tascam LED on the 2408mk3 will blink if a connected Tascam device is not in sync.

Clicks and pops under word clock sync
Many problems result from incorrect word clocking. It is essential that all digital devices in the system be word locked. Consult chapter 3, "Installing the PCI-424 Hardware" (page 25) for detailed information on how to word clock your gear. Whenever there is any weird noise or distortion, suspect incorrect word lock. Tip: the Cuemix Console provides messages helpful for diagnosing problems and optimizing system performance.

Clicks and pops due to PCI bandwidth problems
If Digital Performer or AudioDesk's CPU meter is not overloaded, but you encounter spiking in the peak indicator, you may be exceeding available PCI bandwidth. This may happen if you have other PCI cards installed and/or multiple MOTU audio I/Os. To remedy this, you can disable unused inputs and outputs in the MOTU PCI Audio Console. Also check the CueMix Console message center for details about PCI bus performance.

Clicks and pops under ADAT Sync
Sometimes, the ADAT sync cable seems to be plugged into the PCI-424 card, and partially works - but it isn't really all the way in because it binds against the side of the bulkhead slot. This can cause clicks when slaved to 424 ADAT. Make sure it is really seated firmly. Connect the ADAT sync before screwing the card down - this will ensure that it is aligned properly.

Clicks and pops in 2408mk3 standalone mode
Any time the 2408mk3 is in stand alone mode, not connected to the computer or the computer isn't turned on, the 2408mk3 is acting as a stand alone converter and has a number of choices for word clock settings. If the 2408mk3 is ever trying to clock to some digital source that isn't present, or there are two sources present that are not word clocked locked together, you have the possibility of click's and pop's or sometimes white noise. This is true in any digital scenario.

If the 2408mk3 is in stand alone mode and you have not assigned it the correct clock setting or any clock setting, it could just be sitting there trying to look for word clock or ADAT clock. The solution is to set the 2408mk3 to internal as the clock source or choose the source you are working with. Let's say you are using the analog I/O and a DAT machine with the 2408mk3. In stand alone mode you could choose internal, meaning the 2408mk3 is clocking internally and is the master, ready for you to send audio to your DAT machine. You could choose SPDIF as the clock source if you have the DAT machine on and connected. The sample rate on the 2408mk3 must match the sample rate of the DAT. There are many possible configurations. There must only be one master!

Clicks and pops due to hard drive problems
If you have checked your clock settings and PCI bandwidth and you are still getting clicks and pops in your audio, you may have a drive related problem. Set your Clock Source to PCI-424 Internal and try recording just using the analog inputs and outputs of your 2408mk3. 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, the disk drivers are outdated, or if you are using a SCSI accelerator that is not optimally configured for working with audio.

Connecting or powering gear during operation
It is not recommended that you connect/ disconnect, or power on/off devices connected to the PCI-424 system while recording or playing back audio. Doing so may cause a brief glitch in the audio.

Digital Time Piece - erratic transport behavior
If you experience erratic transport behavior when using Control Track sync from a Digital Timepiece, be sure that the ADAT sync connection to the PCI-424 is not plugged in. ADAT sync to the PCI-424 is not required when using control track sync and may interfere with DTP operation.

Yamaha 02R/03D - Yamaha TDIF card and the 2408mk3
You must have Yamaha's version 2.0 TDIF card which is identified by "CD8-TDII" indicator on the outside of the card. Contact Yamaha's service department at (714) 522-9000 if you do not have this card. Set DIP switch #8 on the card to "DA-88 EXT."

If you are connecting Tascam units like the DA-38/88/98 via TDIF to the 2408mk3 along with a TDIF connection from a Yamaha TDIF card, be sure to put the Tascam unit into Bank A with the TDIF card connection to bank B or C. If you are not connecting a Tascam device, it doesn't matter which bank the TDIF card connection goes to.

PCI Audio Console is blank
The bottom half of the MOTU PCI Audio Console will appear blank if the PCI-424 card is not communicating with your audio interface. When the interface is communicating with the PCI-424 card you should also see the Clock, Source, and Bounce LEDs glowing, or just the Meter led glowing solid, not blinking.

To establish communication set the clock source to internal and hit refresh. Try restarting the computer if refresh doesn't regain communication. Make sure the Audio Wire is plugged in properly. If you still can not regain communication, try one of the other Audio Wire connections on the PCI-424 card and then hitting refresh. Try reseating the PCI-424 card in the computer or even try another PCI slot. If you still have trouble you should contact MOTU Technical Support for further assistance.

Stand alone conversion with the 2408mk3 interface
With the computer off you can configure the 2408mk3 as a stand alone converter. Here is an example for use with and ADAT device on Bank A.

Set the Source Row to ADAT GRP A Set the Clock Row to SRC and it should lock to the sample rate of the source. Set the Bounce Row to 1-2 >1 -2.

Now all inputs coming into Bank A will be routed to all outputs on every Bank.

No inputs available in host application
Check to make sure you have the desired inputs and/or outputs enabled in the MOTU PCI Audio Console. The inputs will be check boxes, and outputs will have pop-up menus. Selecting From Computer will enable an output to be used by your audio application.

Interface banks not visible in the PCI Audio Console
This would indicate that there is a communication problem between the PCI-424 card and the audio interface. Check your Audio Wire connection to make sure it is seated firmly at both ends. To reestablish communication, open the PCI Audio Console, set the clock source to PCI-424 internal, and press the refresh button. If this fails, try connecting the interface to one of the other Audio Wire connections on the PCI-424 card and repeat the above procedure for reestablishing communication.

PCI Audio Console - Can't locate the SPDIF input
SPDIF is only available on Bank C of the 2408mk3. Configure Bank C for SPDIF in the MOTU PCI Audio Console and make sure that the input is enabled.

Monitoring - How to 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 Cuemix Console application that is installed with the drivers. Please see “CueMix DSP hardware monitoring” on page 108.

Noise on shutdown with a 2408mk3

When you quit your audio software, or shut down your computer, your interface will switch to stand alone mode. In stand alone mode, the 2408mk3 is acting as a format converter. If the 2408mk3 is set to an invalid clock source, you may get noise on the outputs of the 2408mk3. To avoid this, make sure that the 2408mk3 has a valid clock source setting in stand alone mode. If you are not sure, set it to Internal.

PCI-424 hardware randomly goes offline

Check that the Audio Wire cable is well seated into the PCI-424 card and the I/O.

Controlling latency

The samples per buffer size setting in the PCI Audio Console, determines the size of the audio buffer before audio is processed. This value also controls the amount of monitoring latency. At its lowest setting, a 64 sample buffer, the monitoring round trip is a mere 128 samples. You can accomplish even lower latency using “Direct Hardware Playthrough” which monitors audio at the hardware level. For further details, see chapter 12, “Reducing Monitoring Latency” (page 105).

Can’t locate inputs and outputs in host software

Be sure you have enabled inputs in the PCI Audio Console. Click the Enable Routing check box to see all inputs and outputs. For details, see “Enable routing” on page 71.

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 complete the registration card included with your PCI-424 system. When we receive your card, you’ll be placed on our mailing list for free software updates and information about new products.

REPLACING DISKS

If your PCI-424 software or AudioDesk CD becomes damaged and fails to provide you with fresh, working copies of the software, 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.

TECHNICAL SUPPORT

If you are unable, with your dealer’s help, to solve problems you encounter with the PCI-424 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
- Tech support email: techsupport@motu.com
- Web site: www.motu.com

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

- The serial number of the PCI424 system. This is printed on a sticker placed on the bottom of the rack-mount audio interface unit. You must be able to supply this number to receive technical support.
- Software version numbers for:

— AudioDesk

— The version numbers of the items in the MOTU Folder (in the Extensions Folder of your startup disk), including the PCI-424 Driver, the MOTU Audio System

- 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 PCI424 system or AudioDesk with which you are having trouble.
- The version or creation date of the system software you are using to run the Macintosh. This can be found by choosing *About this Computer* from the Apple menu.

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.

Our technical support department is dedicated to helping registered users solve their problems quickly. In the past, many people have also taken the time to write to us with their comments, criticism and suggestions for improved versions of our products. We thank them; many of those ideas have been addressed in our development efforts. If you have features or ideas you would like to see implemented, we'd like to hear from you. Please write to the PCI-424 Development Team, Mark of the Unicorn Inc., 1280 Massachusetts Avenue, Cambridge, MA 02138.