MOBILE))I/O
User's Guide

Mobile I/O User's Guide – Table of Contents

Mobile I/O User's Guide – Table of Figures

Mobile I/O: Introduction and Overview

Thank you for purchasing Mobile I/O™, the ultimate FireWire®–based professional audio interface. Mobile I/O is the world's first, and only, portable modular processing audio interface. Your Mobile I/O provides an array of functions that allow you to record and mix with unprecedented quality – Anywhere, Anytime.

This manual covers all operations and features that are common to the Mobile I/O 2882 and 2882+DSP. Items specific to 2882+DSP operation will be available in a separate Users Guide.

What it is

Mobile I/O is a portable, high–quality, modular FireWire–based multi-format audio converter, interface, and processor for professional audio applications. The 2882 model line is equipped with eight balanced analog inputs (4 on XLR and 4 on 1/4" TRS), two channels of Digital I/O (AES/EBU and S/PDIF), eight channels of ADAT® optical I/O support, and eight balanced analog outputs (1/4" TRS), as well as wordclock in/out and 2 IEEE 1394 FireWire connectors that support 400 Mbs operation. All inputs and outputs are capable of 24-bit/ 96kHz operation.

What it has

- 18 simultaneous input channels and 20 simultaneous output channels
- full 24 bit/96kHz audio
- 8 independent channels of mic-pre with switchable phantom power
- Fully Portable Capabilities – Bus and Battery Powerable
- Rack Mount Kit
- 44.1, 48, 88.2, 96kHz Sampling Rates
- 24 bit 110 dB Dynamic Range A/D converters
- 24 bit 120 dB Dynamic Range D/A converters
- 8 Balanced Analog Inputs – 4 XLR, 4 TRS 1/4"
- 8 Balanced Analog Outputs – 8 TRS 1/4"
- Selectable stereo Digital Inputs AES/EBU or S/PDIF
- Stereo Digital Outputs AES/EBU and S/PDIF

- 8 ADAT® Optical Inputs and Outputs
- Sample Rate Conversion (SRC) on Digital I/O
- Built-in 80-bit, fully interpolated, multi-bus mixer for near-zero latency foldback of all input channels and all DAW busses simultaneously
- Full cross point router for I/O management
- Word Clock 1x, 256x
- Front Panel Metering for Analog Inputs and Outputs
- Full console metering of every channel and mix bus
- Total recall of every console parameter

What you need to use it

- Computer:
  - a Power-PC based Macintosh with a FireWire Port and OS 9.1 or newer (B&W G3, G4, Powerbook G3 or G4 recommended). Mac OS 9.2.2 recommended.
  - 128MB of RAM
  - a monitor that supports 1024x768 resolution or better
- Peripheral FireWire Adaptor:
  - OHCI compliant PC-Card or
  - OHCI compliant PCI card
- Software:
  - an ASIO 2 compatible host (such as Cubase, Nuendo, Logic Audio, Digital Performer, Deck or Peak)
  - for ASIO direct monitoring-based foldback and overdubbing, an application that supports ASIO direct monitoring

What comes with it

Your Mobile I/O package contains the following items:

- One Mobile I/O unit
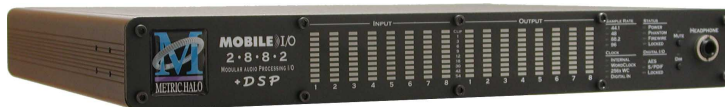
Figure 1: Mobile I/O Unit
+DSP model shown

- One IEC Power Cord appropriate for your area

Figure 2: IEC Power Cord

- One 24-volt 48-watt world-ready external power supply

Figure 3: External Power Supply

- One 0.5 meter IEEE 1394 6-pin FireWire Cable

Figure 4: 0.5 meter 6-pin 1394 cable

- One 4.5 meter IEEE 1394 6-pin FireWire Cable

Figure 5: 4.5 meter 6-pin 1394 cable

- Two Rack Ears w/ fasteners

Figure 6: Rack Ears

- MIO Software CD-ROM
- This Mobile I/O Users Manual
- Warranty/Registration Card

If any of these items are missing from your package when you open it, please contact Metric Halo or your dealer immediately for assistance.

Acknowledgements

Mobile I/O is the result of a tremendous amount of work to deliver a world-changing product. Mobile I/O would not be what it is if not for the efforts of a brave few who worked tirelessly and selflessly to test the hardware and software. We especially want to thank the folks who did focused beta testing on the early releases of the hardware and software and found and helped to resolve a number of nasty issues:

Ed Abbott, Tom Cowland, Orren Merton, Mo Jen, Marco Scocchera, Jon McBride, Hiro Honshuku, Henry Robinette, David Walters, John Leonard, Brian Peters, David Das, Darren Gibbs, Mark Ernestus, Craig Shepard, Al Evans, Steven Campbell Hilmy, Daniel Courville, Jeff Taylor, Matt Mora, Chris Frymire, Louis Chiyoda, Sean Witters, Erlend Myrstad, Floris van Manen, and John Rossero.

Support Information

Warranty

The Mobile I/O hardware is covered by a manufacturer's warranty against manufacturing defects. The details of this warranty are described in a separate enclosed document. This warranty is applicable to products purchased in the USA. Products purchased in other regions are covered by a warranty administered by the distributor for that region and the terms will be as set forth in a separately enclosed document.

Registration

In order to receive warranty service, you must register the product with Metric Halo. This may be done at any time with proof-of-purchase, but we strongly recommend that you register with Metric Halo as soon as you purchase your unit. There are a couple of practical reasons for this:

1. Your product will be registered with us and this registration can be used as proof of ownership if your product is ever lost or stolen.
2. Metric Halo aggressively updates Mobile I/O on a regular basis and we will keep you informed of updates as they become available.

In order to register your Mobile I/O you can send in the enclosed registration card via mail or fax:

Mail Address:

> *Mobile I/O Registration*
> *Metric Halo*
> *Building 8 M/S 601*
> *Castle Point, NY 12511 USA*

Fax Number:

> *+1 (603) 250–2451*

Alternatively, if you have internet access you can use our automated registration webpage at:

http://www.mhlabs.com/mio/register

Service and Support

If you have problems configuring or using your Mobile I/O and you need help, please contact our support group. We offer support via phone and email, as well as peer support on the Mobile I/O Users Group mailing list.

For phone support, contact us at:

+1 (845) 831–8600 x 203

For email support, send email to:

support@mhlabs.com

To subscribe to the Mobile I/O Users Group mailing list or to peruse the archives go to:

http://mail.music.vt.edu/mailman/listinfo/mobileio

Finally, you can always find the latest info and updates for Mobile I/O at the Metric Halo website:

http://www.mhlabs.com/mio

Safety Compliance

This equipment has been tested and found to comply with the limits for a class B digital device pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the 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

- Consult your dealer or experienced radio/television technician for additional assistance.

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.

Using the Mobile I/O Hardware

Mobile I/O Front Panel

Figure 7: Mobile I/O Front Panel

The Mobile I/O front panel provides ten-segment metering for the 8 analog inputs and outputs. The meters are fast PPM peak reading meters with auto-resetting peak holds.

The front panel also provides Mobile I/O system status at a glance:

- Sample Rate (nominal 44.1, 48, 88.2, or 96)
  - The Sample Rate is determined from the currently selected clock source so it will accurately indicate the current sample rate, even when the clock source is being provided by an external device.
- Clock source:
  - **Internal** indicates that the system is internally clocked
  - **Wordclock** indicates that system is being clocked from the wordclock input
  - **256x WC** indicates that the system is being clocked from a 256x clock at the wordclock input
  - **Digital In** indicates that the system is being clocked from the selected digital input (AES or S/PDIF)
  - **256x WC + Digital In** indicates that the system is being clocked from the ADAT optical input
- Power:
  - Indicates that the Mobile I/O is receiving power.
- Phantom power
  - Indicates that at least one of the preamps has the phantom power enabled.

- FireWire
  - Indicates that the Mobile I/O has been successfully connected to a FireWire bus and has detected the isochronous cycle required to transmit and receive audio.

- Locked
  - Indicates that the system clock recovery circuit is properly locked to the selected clock source. If this light is not illuminated, the Mobile I/O will not be locked to a clock and will revert to its failsafe internal clock source. Even if the Locked light is not illuminated, the actual sample rate will still be indicated on the front panel display.

- Digital I/O Section:
  - The AES and S/PDIF lights are mutually exclusive and indicate which of the the two input ports are feeding the Stereo Digital input of Mobile I/O. The Locked light indicates when the digital receiver is locked to the incoming digital audio signal.

The Mobile I/O front panel also provides access to the Headphone output and some associated controls. The headphone output jack is a TRS 1/4" jack that provides the Left Channel on the tip, the Right Channel on the ring and the ground return for the two channels on the sleeve. These signals are all ground referred, so they may also be split and fed single-ended (unbalanced) to an external audio device.

The Mute and Dim buttons provide instant access to simple level control for the headphone output. The Mute button provides a quick, tactile "panic switch" which mutes the front panel headphone outputs in case of accidental feedback loops and other audio unpleasantries. The Dim button attenuates the front panel headphone output by 18 dB.

Mobile I/O Rear Panel

Figure 8: Mobile I/O Rear Panel

The Mobile I/O rear panel features:

- 4 channels balanced XLR inputs. Each input has:
  - 24-bit 96kHz A/D converters (110dB SNR)
  - remote controllable pre-amps with 40 dB of gain
  - remote switchable input impedance characteristics
  - remote switchable 20dB pad
  - remote switchable 48v Phantom power, with 10mA current limit
- 4 channels balanced TRS inputs. Each input has:
  - 24-bit 96kHz A/D converters (110dB SNR)
  - remote controllable pre-amps with 40 dB of gain
  - remote switchable input impedance characteristics
  - remote switchable 20dB pad
  - remote switchable 48v Phantom power, with 10mA current limit
- 8 channels balanced TRS outputs. Each output has:
  - 24-bit 96kHz D/A converters (120dB SNR)
  - remote controllable output gain (from -12dBV up)
- 8 channels of ADAT® lightpipe input (with SMUX support)
- 8 channels of ADAT® lightpipe output (with SMUX support)
- Wordclock input/output on BNC connectors
- 256x Wordclock input/output on BNC connectors
- Stereo S/PDIF input/output on RCA connectors
- Stereo AES/EBU input/output on XLR connectors
- 2 IEEE 1394 (FireWire) ports (400 Mbs)
- 1 2.1mm DC power jack (9v - 30v, center positive, 15 Watts)

Making connections to the Mobile I/O

There are six classes of connections you can make to the Mobile I/O hardware:

1. Analog Audio
2. Copper-based Digital Audio
3. Optical-based Digital Audio
4. Clock Sync
5. FireWire
6. Power

ANALOG AUDIO CONNECTIONS

The analog I/O connections on the Mobile I/O have been engineered for maximum flexibility in that they support both balanced and unbalanced connections with a wide range of input and output levels and a wide range of matching impedances. This means that Mobile I/O handles sources from mic level to line level and from mic impedance to guitar impedance. With that in mind, there are a number of aspects of the design that you should take into account when interfacing with Mobile I/O.

Whenever possible, use balanced interconnects with Mobile I/O. The performance of balanced interconnects is much higher and much more resistant to noise interference and electrical (power) wiring problems. The expense of balanced interconnects is not substantially higher than unbalanced connections, so if the gear that you are interfacing with supports balanced connection — use it.

If you cannot utilize balanced interconnects, there are connection schemes that you can use that will maximize performance.

On input, at line level, it is sufficient to simply use standard unbalanced (TS) connections. If you are interfacing with the Mobile I/O XLR inputs, you will need to ensure that pin 3 is grounded in the unbalanced adapter cable. More information about adjusting the input level can be found in the MIO Console software chapter.

The Mobile I/O XLR inputs are all wired pin 2 hot and the 1/4" inputs are wired Tip hot.

TIP: To use the Mobile I/O TRS input with guitar or bass, you can simply use a standard TS guitar cable (patch cord) and it will work fine. However, you can take advantage of the balanced input design of the Mobile I/O to get more noise rejection than you thought possible on a guitar input. In order to do this, you will need to make a psuedo-balanced telescoping shield guitar cable. This can be constructed with a TRS connector, a TS connector and balanced microphone cable. This cable will treat the guitar as a floating balanced source and provide a telescoping shield from the Mobile I/O ground.
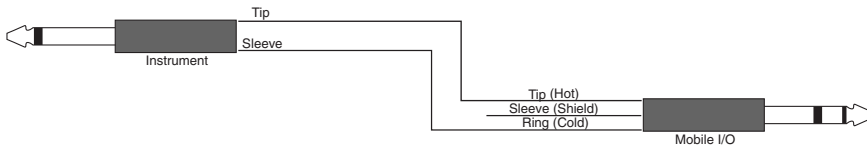
Figure 9: Telescoping Shield Cable for Instruments

If you want to use any of the 4 TRS inputs with balanced (phantom powered) microphones, you will need an XLR female to 1/4" TRS balanced plug adapter cable. These are available commercially, or you can construct one easily. The connections are Tip to Pin 2, Ring to Pin 3 and Sleeve to Pin 1:
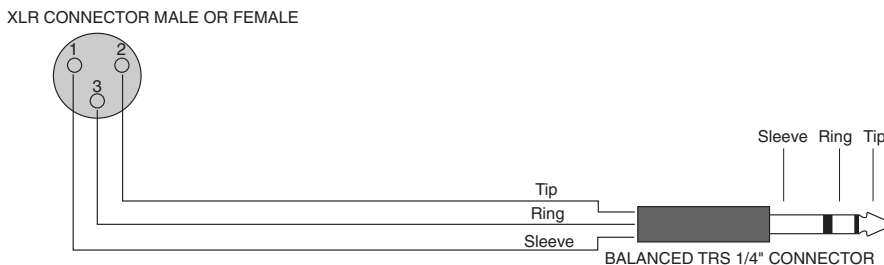
Figure 10: XLR to Balanced TRS Cable

On output, the situation is a bit more complex. If you are driving an unbalanced load, you will get the best performance by not connecting the ring of the TRS jack to ground. In order to do this, you can simply use a balanced TRS/TRS connector with the unbalanced gear. You can also construct a spe-

cial cable with a TRS connector and a TS connector. In this cable, you just let the ring of the TRS connector float:

BALANCED 1/4" TRS CONNECTOR TIP

SLEEVE (SHIELD)

NO CONNECTION

TIP UNBALANCED 1/4" TS CONNECTOR

SLEEVE (SHIELD)
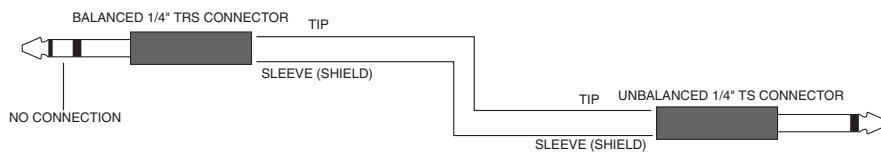
Figure 11: TRS to TS unbalanced cable

Alternatively, the TS connector can be replaced with an RCA connector for interfacing with gear that has RCA unbalanced interconnects.

MAKING THE 1/4" CONNECTION
When you connect a 1/4″ plug to a Mobile I/O jack, insert it straight and firmly, ensuring that the plug is fully inserted into the jack. If the plug is not fully inserted you will get level shifts, phase flips, distortion, or no sound.

To disconnect a 1/4″ plug, firmly pull the plug straight out from the connector body. The connectors on Mobile I/O are stiff, so you may have to exert some force to remove the plug.

MAKING THE XLR CONNECTION
When you connect a Male XLR plug to a Mobile I/O jack, ensure that you have aligned the pins with the connector body and insert firmly until the retention tab clicks.

To disconnect the plug, press the metal retention tab flush against the box, and pull the plug from the Mobile I/O.

COPPER-BASED DIGITAL AUDIO
Mobile I/O supports 2 channels of digital audio over copper-based connections. These connections can be made using either S/PDIF interconnects with the RCA connectors or with AES interconnects using the XLR connectors. Even though only one of the AES or S/PDIF inputs can be active at any given time, you can have different digital sources connected to each of the input connectors at the same time – you use the MIO Console application to select the active input. Audio routed to the digital outputs will be mirrored

by both S/PDIF and AES outputs. This allows you to send the same stereo pair to two devices at once.

We recommend that you use the AES interconnect mechanism to establish the digital communication between the Mobile I/O and other digital devices. The jitter and electrical noise tolerance on AES interconnects is substantially better than with S/PDIF interconnects. The AES interconnect standard is equivalent to balanced audio interconnections. If you need to use S/PDIF interconnects, try to use the shortest cables you can and, if possible, use special purpose 75 ohm S/PDIF or video cables.

The RCA connectors used for S/PDIF are friction fit coaxial connectors. When you connect them, ensure that they are fully inserted and tight.

The XLR connectors used for AES are fully locking. When connecting to them, make sure that you align the pins and insert firmly. When you remove the connector, make sure that you release the lock by pressing the lock release button before you pull the connector out of the Mobile I/O.

INTEGRATED SRC

Normally, when working with digital audio transport, you must take care to ensure that all devices communicating with one another are synchronized to the same audio clock. While this is still an important consideration with Mobile I/O, the hardware provides a special feature to simplify copper-based digital connections to the box. The digital input on Mobile I/O has an optional asynchronous sample rate converter (SRC) that will automatically match the sample rate of the incoming audio to the sample rate of the Mobile I/O. This converter is enabled by default and you can disable it in the System section of the MIO Console. If you have synchronized the Mobile I/O to the external source (using any of the extensive synchronization methods provided by Mobile I/O), you will generally want to disable the SRC in order to get 24-bit transparent audio transport over the digital input.

OPTICAL-BASED DIGITAL AUDIO

Mobile I/O provides two TOSLink™ connectors on the back panel. One is a transmit connector and the other is a receive connector. These connectors are used with Plastic Optical Fiber (TOSLink) cables to communicate with

other devices that support the ADAT® Optical communication protocol. The ADAT Optical standard allows a device to transmit 8 channels of 24-bit audio at up to 50kHz along with digital audio clock information.

After the original 8 channel specification was finalized, Alesis and other third parties extended the standard to support 4 channels of 24-bit audio at 2x sample rates. This extension of the standard is commonly referred to as SMUX. Mobile I/O automatically supports SMUX transport over the ADAT Optical connections when the box is in 2x mode (either 88.2k or 96k sampling frequency). While operating in SMUX mode, the audio data for each of the first four "ADAT" channels is placed on pairs of optical channels. ADAT 1 is placed on optical channels 1 and 2. Each optical channel continues to transmit data at 1x speeds, and both devices are required to multiplex and demultiplex the audio into and out of the optical channel pairs.

When Mobile I/O is receiving audio over the ADAT optical input, you will generally get the best results when you have selected ADAT as the audio clock source for Mobile I/O. You may find that other clocking combinations work reliably, but the reliability of the the ADAT input will depend on the details of the external device's clocking circuitry.

Since Mobile I/O provides direct routing within the box, you can easily configure the unit to work as an ADAT based 8 channel A/D/A. Refer to the chapter on MIO Console for information about configuring the patchbay router.

CLOCK SYNC

Clock sync is a serious consideration in any digital audio system.

If you are recording analog sources with Mobile I/O, you can simply use the unit's high-quality internal clock source to drive the converters. This is the easiest case to deal with.

If you need to interface with other devices digitally or ensure sample accurate sync with video sources, the extensive clock synchronization capabilities of Mobile I/O will prove to be more reliable (and better sounding) than most higher priced alternatives.

There are four different ways to get external clock information into the unit:

1. Sending a 1x word clock signal into the WC Input BNC.
2. Sending a 256x word clock signal into the WC Input BNC.
3. Sending an AES or S/PDIF signal into the Digital input.
4. Sending an ADAT optical signal into the Optical input.

The BNC word clock input port is a 75 Ohm terminated coaxial input. It should be driven by a 75 Ohm source driver and interconnected with 75 Ohm coaxial cable. If you do not use proper cabling and source drive, you will introduce reflections on the word clock cable which will propagate jitter into the recovered word clock. This is true whether you use the port as a 1x WC input or a 256x WC input, but becomes more important when the clock signal is 256x.

1x is generally appropriate for use with devices that provide a word clock output. If your device provides a 256x output, you may find that you get better results using that clock signal. The Digidesign® line of Pro Tools® products use 256x as their "Superclock™" clocking signal.

The AES recommended procedure for distributing clock is to use an AES clock signal. The AES clock signal is an AES digital audio signal with no audio activity. Mobile I/O only uses the AES preambles for clock recovery, so it is immune to data dependent jitter effects. This means you can reliably use the Digital Input as a clock source with or without audio data.

Finally, you can recover clock from the optical input sent to the Mobile I/O from another ADAT device. In general, if you want to use the ADAT optical input to receive audio, you will want to choose the ADAT optical input as the clock source.

FIREWIRE

FireWire® is Apple's registered trademark for the IEEE 1394 High-Speed Serial Bus. FireWire started as an Apple technology to replace a variety of interface ports on the back of the computer. After promulgating a number of closed proprietary technologies in the early days of the Macintosh, Apple determined that open standards were better for the Mac, for the industry,

and for Apple itself. On that basis they opened their technology for standardization under the auspices of the Institute of Electrical and Electronics Engineers, Inc. (IEEE), an international organization that promotes standards in the field of electronics. FireWire was standardized as IEEE 1394 and promoted for open licensing in the industry.

The first widespread adoption of the technology was for DV camcorders where space was at a premium and bus powering was not percieved as a real issue since all camcorders have batteries. Sony designed an alternative version of the standard 6-pin FireWire connector that provided 1394-based communication with 4-pins in a much smaller form-factor. This version of the connector sacrificed bus-power support and mechanical stability for reduced space requirements. Sony dubbed this version of IEEE1394 "i.Link®." This became the de facto standard in the DV world, and was later added to the IEEE 1394 standard. Both i.Link and FireWire refer to the same underlying standard and are completely interoperable. Obviously, i.Link connectors and FireWire connectors cannot be used together without adapters.

Mobile I/O uses the FireWire flavor of the IEEE1394 connector with 6-pins for bus power support. The unit ships with two 6-pin to 6-pin FireWire cables, one that is 0.5 meters long (about 18 inches), and the other 4.5m (about 14.5 feet) long. If you want to use Mobile I/O with a 4-pin FireWire device, you will need to purchase a 6-pin to 4-pin adapter cable. These cables are available from a wide variety of retail sources. If you are using a 4-pin cable to connect any device to the computer with Mobile I/O, bus power will not be available.

The 6-pin FireWire connector is polarized by its shape, one end of the connector is pointed. The FireWire ports on Mobile I/O point downwards toward the bottom of the box. It will be very difficult to insert the connector upside down, but it is possible if you force it. If the plug is inserted into the socket upside down, the socket will be destroyed. **NEVER FORCE A FIREWIRE CONNECTOR INTO A FIREWIRE SOCKET.**

Devices connected to the FireWire bus are autoconfiguring. You do not need to set IDs or DIP switches or in any way configure the devices in order to facilitate communication between devices or to configure of the bus.

FireWire devices on the same bus must be connected in a tree structure with no loops. This means that devices can be connected to each other in any order, and any device with multiple ports can act as a chain or a hub for other FireWire devices, but you should never be able to get from one device to another by more than one path. If you construct a loop in the bus, it will not operate properly and you will not be able access some or all of the devices on the bus.

Although you are able to attach devices in any order on the FireWire bus, the order of attachment will have an impact on performance. Most current model FireWire devices support 400 Mbs operation, but many older devices may only support 100 or 200 Mbs operation. These devices act as a bottleneck in the bus and limit the speed of any bus traffic that flows through them. In order to maximize performance, you want to ensure that low speed devices are not used to join high speed devices. In practice this generally means that you should attach your Mobile I/O directly to your computer or through a high speed hub.

To connect Mobile I/O to your computer simply plug a FireWire cable into the Mobile I/O and into the computer. The FireWire bus provides a path for all communication between the computer and Mobile I/O – audio, control and meter data.

Mobile I/O audio transport takes advantage of FireWire's support for isochronous transmission, in which the Mobile I/O can reserve a dedicated amount of bandwidth on the bus for moving audio samples. Since the audio **must** be transmitted on a regular basis to ensure continuous playback and recording , the isochronous model is perfect.

Control changes and meter data are transmitted using asynchronous transactions on the FireWire bus. This transmission approach makes use of the unreserved bandwidth on the bus and competes with things like FireWire hard disk accesses for time. Under normal circumstances this is completely transparent to the user. If the bus becomes overloaded, you may find that disk

accesses and meter updates slow down. If you are experiencing bus over-loads, you can always add a second FireWire bus with a third-party FireWire card (PC-Card or PCI card depending on your machine), and offload one or more devices to the second bus.

POWER

One of Mobile I/O's great strengths is the flexibility of its power system. Mobile I/O can be powered from any DC source (including bus power) in the range of 9V to 30V as long as it provides 12 Watts of power. The DC input on Mobile I/O is a 2.1mm coaxial power connector, center positive. So if you are powering the unit with a third party power source and it supplies 9V, the power source will have to provide 1.4 amps of current. If you are powering the unit with 12V, the power source will have to provide 1 amp of current, and so on.

The Mobile I/O ships with a world-ready 24 volt, 2 amp power supply. You can plug this supply into any AC power source from 90V to 240V, 50Hz - 60Hz, using an appropriate IEC power cord, and it will supply the proper power to the Mobile I/O on the 2.1mm coaxial power connector. Mobile I/O will automatically supply the extra power to the FireWire bus. This means that the Mobile I/O and its power supply can be used to power other bus-powerable FireWire devices including hard-drives, hubs, and other Mobile I/O units.

Since Mobile I/O is DC powered, you can also power up the Mobile I/O using the FireWire bus or another DC source. The Mobile I/O uses 12 Watts of power, so the device supplying the bus power must be capable of sourc-ing that much power. Most desktop Macs provide more than enough power for Mobile I/O and one other low power device. Most laptops provide enough power for Mobile I/O, but not enough for Mobile I/O and another bus-powered device at the same time. If you are using a Powerbook com-puter, you should not expect to be able to power both the Mobile I/O and a hard drive from the computer. The power capabilities of individual comput-ers vary, so you will have to test the complete system to determine exactly how much your computer can handle.

If you find that the computer is not capable of powering Mobile I/O or does not provide enough run time, you may want to explore using an external power source with the Mobile I/O. Check with Metric Halo for details on different battery power solutions for Mobile I/O.

As with all electronic devices, when connecting an external power source to the Mobile I/O, you should first connect the power source to Mobile I/O while it is in an unenergized state (e.g. not connected to the mains or switched off). After the connection to Mobile I/O has been made, you should energize the power source.

If you connect an energized power source to the Mobile I/O's 2.1mm power connector you may see a small spark when you make the connection. This is due to surge current and is normal if you connect a power source in this way. While this will not damage the Mobile I/O in any way, to avoid the spark just connect the power connector to Mobile I/O before connecting the power source to the wall.

MIO Console
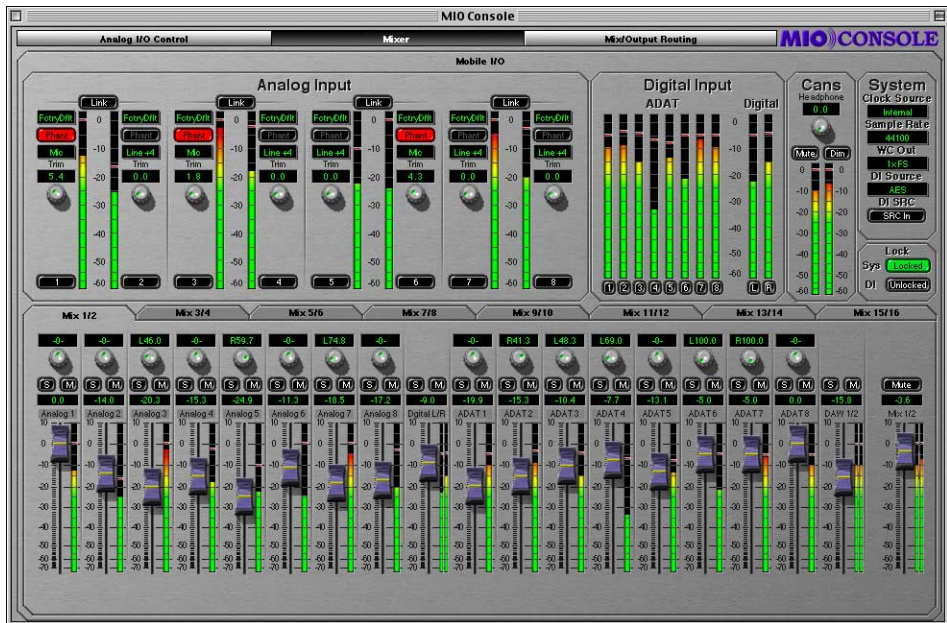
Figure 12: MIO Console

MIO Console is the nerve center of your Mobile I/O. Functioning as a standalone application, MIO Console provides full control of every aspect of Mobile I/O. The console software allows you to rapidly and easily adjust all of the Analog Input and Output channel parameters, system sample rate, Digital I/O source, and system clock source. It also allows you to assign ASIO output channels and hardware input channels to the integrated 80-bit, fully-interpolated, multi-bus, near zero-latency hardware mixer.

MIO Console's powerful mixer supports both mono and stereo busses, with solo and mute functions for all input and master channels. The mixer bus outputs are routable to any of the hardware outputs, allowing you to easily create multiple simultaneous mixes for send/return busses and multiple live main and monitor mixes. Various applications include foldback support for multiple performers, separate monitor feeds for studio, tape, and control room, and separate mixes for front of house, archive recording, and monitors for live shows.

MIO Console also contains a patchbay router, which allows you to quickly select the source being fed to any output. The patchbay provides easy configuration of stand-alone operation, mix mults, direct outs and various combinations thereof to suit the needs of the moment.
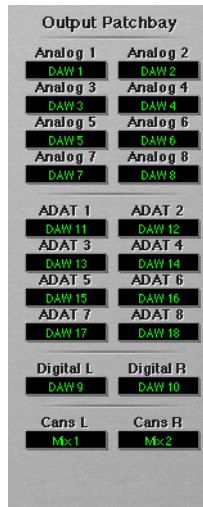
Figure 13: MIO Console Output Patchbay

In order to simplify work flow and optimize the extent of system control, MIO Console supports comprehensive preset management on both a global and individual control level.

The preset management popup controls within MIO Console allow you to configure various aspects of Mobile I/O and save that configuration information for later recall. Various applications include storing routing configurations for monitor setups, mixer configurations for stem and scene recall, and storing analog level standards for interfacing with external gear and managing different mastering standards.

Global configuration snapshots allow you to save each and every aspect of Mobile I/O's configuration for later, total instant recall. This is useful for pre-configuring Mobile I/O and bringing back the configuration once at the gig, managing separate location setups, or for saving complex studio routing setups for quick changeover.

MIO Console Overview

The MIO Console application consolidates all of the controls for the Mobile I/O hardware into one easy to use window. The Mobile I/O has an extremely large number of user configurable parameters and it is very important that you have instant access to the ones that you need. The console provides a thinly layered interface to the entire system and keeps you from having to deal with "Window Overload".

The MIO Console window has a **view panel selector bar** that runs along the top of the window. This bar indicates which of the console view panels is currently active. You can tell which panel is active because the button in the bar is "pushed in." To switch to one of the other panels, simply click on the name of the panel you want to use. The view will change instantly to the one that you have selected.

Figure 14: View Panel Pane Selector Bar

Under the **view panel selector bar** is the currently selected view panel. You control the various aspects of the box with the controls in each view. The MIO Console has three main panels:

1. Analog I/O Panel
2. Mixer Panel
3. Routing Panel

ANALOG I/O VIEW
This panel provides full control and metering of all of the analog I/O that the box provides. The top half of the view is dedicated to inputs and the bottom half is dedicated to outputs. You access this panel by clicking on the "**Analog I/O Control**" button of the **view panel selector bar**.

ANALOG INPUT CONTROL
For each analog input channel on the Mobile I/O, you will find a channel strip that contains:

23
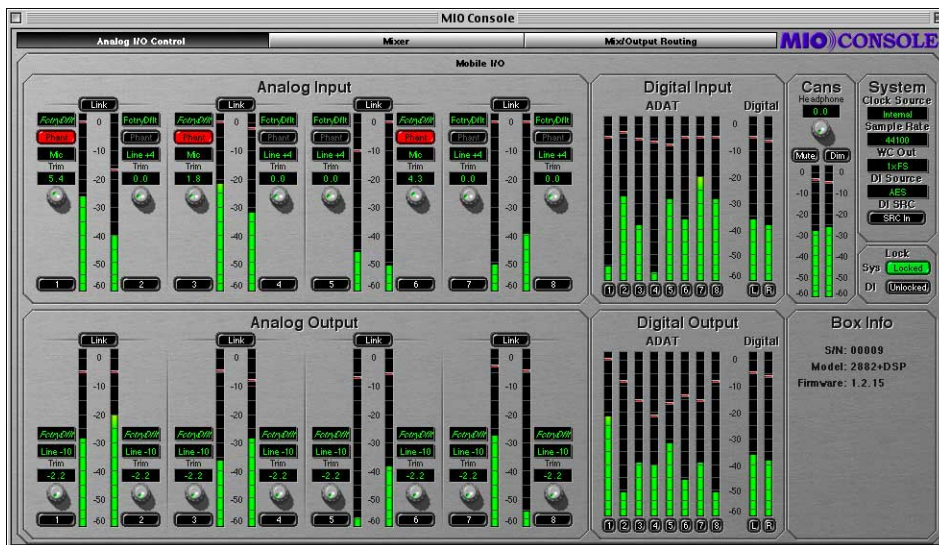
Figure 15: Analog I/O Panel

1. **Parameter Popup** control menu

Figure 16: Parameter Pop-up Menu

• The Parameter Popup control allows you to save, recall, and manage all of the parameters associated with a channel strip in one place. The control is documented in detail later in this chapter (see: "**Parameter Popup Controls**" on page 50). All presets are automatically shared between all of the input channels.

2. **Phantom Power** enable button

Figure 17: Phantom Power Button

• The Phantom Power enable button allows you to control whether or not +48v phantom power is applied to the input by the Mobile I/O. This button is only enabled if you have selected Mic or Mic/Pad as the input level

standard. If you are using one of the other level standards, Phantom Power is automatically disabled. If you have enabled Phantom Power on this channel, the button will be illuminated bright red. In addition, the Mobile I/O front panel will illuminate the "Phantom" indicator if any of the input channels have Phantom Power enabled.

• Phantom power is appropriate for use with condenser microphones or other devices that can (and must) be powered by the preamp. Mobile I/O limits the amount of phantom power to 10mA per channel, preventing device damage due to shorts. Some (rare) microphones may require more power than is provided by 10mA; you will need an external power supply to power those mics.

3. **Level Standard** popup menu

Figure 18: Level Standard Pop-up Menu

• This control allows you to select the input level and impedance characteristics for the input channel. The available choices are:

• Line +4

This format supports input levels up to +26 dBu. The input impedance is approximately 10k, and the inline pad is engaged. Phantom power is not available.

This format is appropriate for interfacing with professional audio equipment.

• Line -10

This format supports input levels up to +26 dBu. The input impedance is approximately 10k, and the inline pad is engaged. Phantom power is not available.

This format is appropriate for interfacing with prosumer and consumer audio equipment.

• Inst (Instrument)

This format supports input levels up to +6 dBu. The input impedance is approximately 200k, and the inline pad is not engaged. Phantom power is not available.

This format is appropriate for interfacing with high output impedance sources like guitar pickups.

• Mic

This format supports input levels up to +6 dBu. The input impedance is approximately 200k (12k with phantom power engaged) and the inline pad is not engaged. Phantom power is available.

This format is appropriate for interfacing with dynamic and condenser microphones on low to mid level sources. You may find that for very low level sources, especially with low-sensitivity microphones, that your SNR improves if you take advantage of a high-quality external mic preamp. External preamps are recommended for certain types of recording and microphones, especially classical recording and ribbon mics.

- Mic/Pad

This format supports input levels up to +26 dBu. The input impedance is approximately 10k, (6k with phantom power engaged) , and the inline pad is not engaged. Phantom power is available.

This format is appropriate for phantom powered condenser mics that put out near-line level signals (often the case when you are using condenser microphones on bass drums or other very loud sources).

4. **Gain Trim** knob

Figure 19: Gain Trim Knob

- The Gain Trim knob allows you to adjust the analog gain of the input stage over a 40dB range determined by the input standard that you have selected. The gain is indicated in dB relative to the nominal level of the input standard you have selected. The gain changes are smoother near the bottom of the scale, with the steps increasing in size as you reach the +40dB gain limit.

5. **Channel Label** button

Figure 20: Channel Label

- This simply labels which channel is associated with the channel strip.

6. **Channel Level Meter**

- This is a peak reading, high-resolution, fast PPM meter. It shows the post converter level of the input signal of the associated channel.. The peak hold bar indicates the highest level seen on the channel since the last

Figure 21: Channel Level Meter

reset. You can reset the hold by clicking on the meter. These meters are simply high resolution versions of the meters shown on the front panel of the box – all the meter data is generated by the Mobile I/O hardware.

OPTIMIZING INPUT LEVELS

The Analog to Digital converters (ADC) in most devices function best when the peak level is around -6 dBFS (lowest distortion, best sound). This is true of the ADCs in Mobile I/O. Since you have full level control of the input with the gain trim knob, you will find that you get the best quality recordings if you try to set the nominal peak level of the input at about -6 dBFS. In addition to providing the best recording quality, it has the added benefit that you will be operating with an extra 6 dB of headroom before clipping. There is no drawback to optimizing your levels in this way, and plenty of benefit.

ANALOG INPUT CHANNEL LINK

Figure 22: Analog Input Channel Link

In addition to the channel specific controls, each channel pair shares a **Link** button. When the Link button is engaged, changes made to one channel of the pair will automatically be applied to the other channel of the pair. This is very useful if you are miking with a stereo pair and need to maintain level balance between the two preamps -- with the Link button engaged, the balance is automatic and exact. When you enable the Link button, the control

values for the odd channel in the pair will be copied to the even channel of the pair.

DIGITAL INPUT METERS

To the right of the Analog Input control section is the Digital Input Meter section. This group of meters provides level metering for all of the digital inputs on the Mobile I/O. These meters have the same response characteristics as the analog input meters and show you the audio activity on ADAT channels 1-8 and digital input channels 1-2, going from left to right. Please note that when the box is clocking at 2x rates (88.2-96k), the ADAT input uses pairs of optical channels to transport the audio, and only channels 1-4 will show activity – channels 5-8 will not display any signal.

CANS CONTROLS

The next control block to the right is the Cans (Headphones) control section.

Figure 23: Headphone Controls

The Headphone level control knob provides about 40dB of analog gain control on the headphone output. The **Mute** and **Dim** buttons reflect the state of the hardware mute and dim controls for the front panel headphone output. Engaging one of these buttons will also engage the associated control on the hardware.

When the **Mute** function is engaged, the headphone output will be muted. When the **Dim** function is engaged, the headphone output will be padded

by 18 dB. The meters in this section show the headphone output pre-mute/dim block.

SYSTEM CONTROLS

Figure 24: System Controls

The **System** block provides controls that adjust various system level aspects of the Mobile I/O hardware:

1. The **Clock Source** popup menu controls the system clock source used by the hardware for digital synchronization and driving the converters:
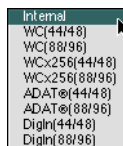
Figure 25: Clock Source Pop-up Menu

- **Internal** causes Mobile I/O to use its internal clock. You must select this if you want to set the sample rate from the Mobile I/O. If any other clock source has been selected, the console will not allow you to change the

29

sample rate since the sample rate is determined by the external clock source.

- **WC (44/48)** directs Mobile I/O to clock off of an external Word Clock Source at single rate (e.g. fs = 32k-50k)

- **WC (88/96)** directs Mobile I/O to clock off of an external Word Clock Source at double rate (e.g. fs = 64k-100k)

- **WCx256 (44/48)** directs Mobile I/O to clock off of an external 256fs Clock Source at single rate (e.g. fs = 32k-50k)

- **WCx256 (88/96)** directs Mobile I/O to clock off of an external 256fs Clock Source at double rate (e.g. fs = 64k-100k)

- **ADAT (44/48)** directs Mobile I/O to clock off of the incoming ADAT stream and run at single rate. All 8 ADAT channels are available. You will generally want to select this source if you intend to use ADAT input.

- **ADAT (88/96)** directs Mobile I/O to clock off of the incoming ADAT stream and run at double rate. The bottom 4 ADAT channels are multiplexed over the lightpipe to provide 4 channels of double rate audio compatible with other SMUX and Alesis 96k ADAT devices. If you are using a device that provides 4 channels of 96k audio over ADAT optical, you will want to select this clock source.

- **DigIn (44/48**) directs Mobile I/O to clock off of the selected stereo digital input at single rate (e.g. fs = 32k-50k). This allows operation of the digital input without SRC, and from devices that must supply clock.

- **DigIn (88/96**) directs Mobile I/O to clock off of the selected stereo digital input at double rate (e.g. fs = 64k-100k). This allows operation of the digital input without SRC, and from devices that must supply clock.

2. The **Sample Rate** popup menu allows you to select the sample rate when you are using internal clock. The Mobile I/O must be running on internal clock for the Sample Rate popup menu to have any effect. If the Mobile I/O is running from an external clock source, you cannot select the sample rate since it is determined by the external clock source.
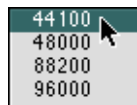
Figure 26: Sample Rate Popup Menu

3. The **WC Out** popup menu allows you to select the output clock signal the Mobile I/O generates on its WC Out BNC connector. The available choices are 1x and 256x. The 1x signal is appropriate for driving devices that accept a Word Clock signal. The 256x signal is appropriate for driving devices that accept 256x or SuperClock signals. Refer to the documentation for the external device to determine what is the most appropriate clock reference for it.

Figure 27: WC Out Popup

4. The **DI Source** popup menu allows you to select the active input for the digital input pair. The choices are AES and S/PDIF. This selector physically switches the input to the digital audio receiver between the RCA input and the XLR input.

Figure 28: DI Source Popup

5. The **DI SRC** button enables and disables the asynchronous sample rate converter (SRC) in the Mobile I/O digital audio receiver. When the SRC is engaged (button illuminated yellow), the digital audio receiver will automatically synchronize the input signal to the Mobile I/O system clock over a wide range of sample rate ratios.

Figure 29: DI SRC Buton

This allows you to, for example, digitally transfer a sample from a CD player into a 96k session without any clocking problems. If you want to make bit-transparent transfers, you will need to disengage the SRC and ensure that the Mobile I/O and the external device are both using the same digital audio clock via one of the Mobile I/O synchronization mechanisms.

6. The **Lock** indicators show which elements of the Mobile I/O clocking system are properly locked. The clocking system must be locked for the unit to behave as expected. If the system is not locked, audio will play at the wrong rate and will be distorted or noisy. Under normal circumstances, the system should always be locked, but if you have selected an external clock source and the

clock signal is not present, corrupted or out-of-range, the system may unlock. There are indicators for the system and the digital input.

Figure 30: Lock Indicators

ANALOG OUTPUT CONTROL

The bottom half of the panel is dedicated to the hardware outputs of the Mobile I/O. Much like the analog inputs, each of the eight analog outputs has a set of controls associated with it. The controls are similar to the analog input controls. Each output channel has:

1. **Parameter Popup** control
   - The Parameter Popup control allows you to save, recall, and manage all of the parameters associated with an analog output in one place. The control is documented in detail later in this chapter (see: "**Parameter Popup Controls**" on page 50). All presets are automatically shared between all of the output channels.

2. **Level Standard** popup menu
   - This control allows you to select the output level for the channel. The available choices are:
     - Line +4

       This format supports output levels up to +26 dBu. The standard setting with the trim knob set to 0dB yields a +24dBu output when the digital signal driving the DAC is 0dBFS. This corresponds to +4 dBu nominal with 20dB of digital headroom. You can use the trim knob to adjust the analog output level to be consistent with the needs of other audio gear. The output impedance is approximately 50 Ohms.

       This format is appropriate for interfacing with professional audio equipment.
     - Line -10

       This format supports a nominal output level of -10dBV with 20 dB of digital headroom. You can use the trim knob to adjust the analog out-

put level to be consistent with the needs of other audio gear. The output impedance is approximately 50 Ohms.

This format is appropriate for interfacing with prosumer and consumer audio equipment.

3. **Gain Trim** knob
   - The Gain Trim knob allows you to adjust the analog gain of the output stage (post DAC) over a 40dB range determined by the output standard that you have selected. The gain is indicated in dB relative to the nominal level of the output standard you have selected. The gain changes are smoother near the bottom of the scale, with the steps increasing in size as you reach the +40dB gain limit. Unless the signal is quite low, you will put the output stage into analog clipping well before you hit the 40dB gain limit.

4. **Channel Label** button
   - This simply labels which channel is associated with the output.

5. **Channel Level Meter**
   - This is a peak reading, high-resolution fast PPM meter. It shows the pre converter level of the output signal of the associated channel. The peak hold bar indicates the highest level seen on the channel since the last reset. You can reset the hold by clicking on the meter. These meters are simply high resolution versions of the meters shown on the front panel of the box – all the meter data is generated by the Mobile I/O hardware.

ANALOG OUTPUT CHANNEL LINK

In addition to the channel specific controls, each channel pair shares a **Link** button. When the Link button is engaged, changes made to one channel of the pair will automatically be applied to the other channel of the pair. This is very useful if you are driving a stereo monitor section or stereo device. By engaging the Link button, you will ensure that both channels have precisely the same amount of analog gain applied post DAC. When you enable the Link button, the control values for the odd channel in the pair will be copied to the even channel of the pair.

DIGITAL OUTPUT METERS

To the right of the Analog Output controls is the Digital Meters section. This group of meters provides level metering for all of the digital outputs on the Mobile I/O. These meters have the same response characteristics as the analog output meters, and show you the audio activity on ADAT output chan-

nels 1-8 and digital output channels 1-2, going from left to right. Please note that when the box is clocking at 2x rates (88.2-96k), the ADAT output uses pairs of optical channels to transport the audio and only channels 1-4 will show activity, channels 5-8 will not display signal. Both channels of the stereo digital output remain active at all sample rates.

BOX INFO

The **Box Info** section of the panel, in the lower right-hand corner of the window, shows you information about the currently connected and selected Mobile I/O unit. This section displays the Serial Number, Model Information and Firmware revision of the connected box. All of this information can be useful in trying to track down any connection problems that may arise.
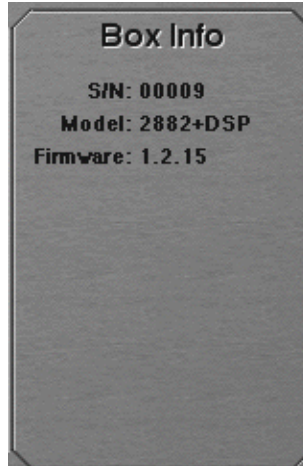
Figure 31: Box Info

If there is no information displayed in the Box Info section, the software is not communicating properly with the Mobile I/O hardware, or there is no Mobile I/O present on the FireWire bus.

If the FireWire light on the front panel of the Mobile I/O is illuminated but the box information does not appear in the console window, it is very likely that the software has not been installed properly. If this is the case, please refer to the installation instructions for details on how to properly install the software. Also, pleaseensure that there are no copies of the MobileIO Driver

in the same folder as the MIO Console application or any of the ASIO drivers that you will use with an ASIO host.

If, on the other hand, the FireWire light on the front panel of the Mobile I/O is not illuminated, the box is not communicating properly with the computer. Please check the cabling of your Mobile I/O and other devices on the FireWire bus and make sure that everything is connected correctly. If that does not properly establish the connection, try rebooting your computer. As a last resort, try connecting only the Mobile I/O to the computer to ensure that communication can be established.

MIXER PANEL

The Analog I/O panel provides all of the bread-and-butter functionality that you would expect from a flexible audio interface like Mobile I/O - what you need to get the job done quickly and easily. The Mixer panel provides features that you won't find anywhere else – here is where things start to get *really* interesting.
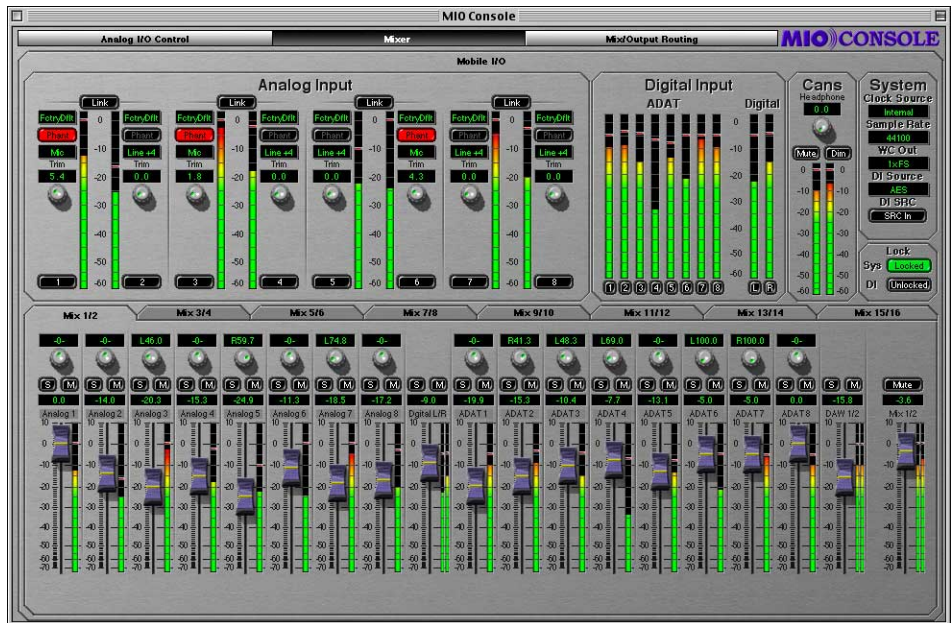
Figure 32: Mixer Panel

The controls in the top half of the mixer panel are the same analog input controls described in the section **"Analog I/O View"** on page 23. The bottom half of the mixer panel provides a control surface for the multiple integrated hardware mixers running on Mobile I/O. You access this panel by clicking on the "**Mixer**" button of the **view panel selector bar** at the top of the console window.

MIXER CONTROLS OVERVIEW

Each mixer has controls for Level (fader), Solo (button), and Mute (button) for each channel. Each stereo mixer also includes a pan control (knob) for each mono channel. For stereo inputs, such as the defaults of the digital inputs or your DAW output, will not have a pan knob. Instead, the two individual channels that make up the pair will be hard panned, and the fader meters will appear as stero meters.

At the far right of each mixer surface is the Master Fader. This fader will be Mono for Mono Mixers and Stereo for Stereo Mixers. Above the Master Fader is a bus mute button that allows you to quickly mute the entire mixer.

Each mix bus can have up to 36 inputs assigned to it. All of the analog, digital, and ADAT hardware inputs are available in each mixer. You can also assign any (or all) of the ASIO playback channels from your DAW to each mixer. Since there are 18 hardware inputs and 18 FireWire return busses that are routed from the ASIO driver on the host computer, you can mix up to 36 channels on each bus.

The default state of the hardware input channels in the mixers is faders at unity, mutes on, solos off and center panned. The playback channels from your DAW are unmuted in pairs per bus (e.g. DAW 1/2 will be unmuted on Mix 1/2, DAW 3/4 will be unmuted on Mix 3/4, etc.)

MIXER PANE TABS

Use the tabs above the pan knobs to select the mixer you want to control. Each tab represents a mixer you have configured in the routing panel. When you click on a tab, the controls will be instantly updated to reflect the state of the associated mixer. In this way you can quickly switch back and forth between multiple independent mixes, each with a full mixing interface.

Figure 33: Mixer Pane Tabs

Each mixer also has a **Parameter Popup** control associated with it. Unlike the Input and Output controls, the parameter control popup does not have an explicit user interface element representing it. Instead, the popup is accessed from the mixer's tab itself. To pop-up the menu, you either click and hold the tab, or <control>-click the tab. The parameter popup menu allows you to maintain a library of standard mix configurations, scenes, and setups (see: "**Parameter Popup Controls**" on page 50). It also provides a very quick method for copying mixes from one bus to another.
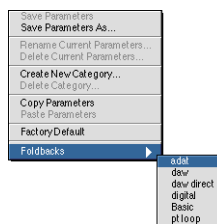
Figure 34: Mixer Parameter Popup

CHANNEL FADERS

The Channel fader controls the relative level of the input channel in the mix. It works just like its non-virtual counterpart. The calibration numbers to the left of the fader knob provide an accurate guide to the amount of gain that will be applied by the fader. The label area above the fader knob lets you know which input channel the fader is controlling. The labels default to the console's default names for the channels, but you can rename the channels to more meaningful names using the naming controls in the Routing Matrix (see: "**Configuring Channel Names in the Matrix**" on page 48).

The exact amount of gain for the fader is displayed in the small black window above the fader channel label. If you want to set a channel gain precisely, simply click in the small black window and type in the desired gain in dB numerically. Hitting the <return> key or clicking outside the entry box makes the new setting take effect. Hitting the <tab> key will make the new setting take effect and will move to the next numeric entry field in the mixer.

The gain range for each channel is from -∞ (muted) to +10 dB. The resolution of the gain control is extremely fine and gain changes are interpolated in the mixer, so there is no zipper noise when you make continuous (or even discontinuous) gain changes.

<Command> clicking the fader knob will allow you to make fine adjustments to the fader level by dragging the mouse up and down. <Option> clicking the fader knob will reset the fader to unity gain. Clicking the meter associated with the fader will clear the peak holds for that channel.

For mono channels, the current gain is applied to the channel before the pan. For stereo channels, the current gain is applied uniformly to both of the hard-panned input channels.

CHANNEL METERS

Each channel fader has a meter (or a pair of meters for stereo channels) associated with it. The meters are calibrated consistently with the fader calibration. Each meter is a peak reading, fast-PPM meter that is pre-fader. The peak hold bar shows the biggest peak since the last reset. Clicking on a meter will clear the peak holds for that channel.

CHANNEL PANS

Each mono input in a stereo mixer has a pan knob above the fader. The pan knob allows you to control the relative amount of the input channel that is placed into the two busses of the stereo mixer. Panning hard left (L100) means that the channel will appear at full volume in the left (odd) bus of the mixer. Panning hard right (R100) means that the channel will appear at full volume in the right (even) bus of the mixer. When the channel is center panned, the signal appears at a decreased volume (-3dB) in both channels, such that the volume of the total signal in both channels is equivalent to full volume in one channel. As you pan from left to right, the signal is distributed between the two channels so that the total volume remains constant.

As with the channel fader, the black window above the channel pan knob provides a precise readout of the current pan position, and clicking on the black window will allow you to type in an exact pan amount. Negative numbers (-100 to 0) indicate left pans and positive numbers (0 to 100) indicate right pans. 0 is center pan.

<Option> clicking the pan knob will return it to center pan, and <command> click-dragging the knob will allow you to adjust the pan position in finer increments.

MUTE BUTTONS

Each input has a Mute button (labeled "M") associated with it. When the mute button is engaged (illuminated white), the channel will be muted in the mixer. The mute is interpolated, so muting a channel will not cause audible clicks.

<Option>-clicking a mute button will set all of the mutes on the mix bus to the same state as the button you click. This allows you to quickly mute all the channels or quickly unmute all the channels on a bus.

SOLO BUTTONS

Each input has a Solo button (labeled "S") associated with it. When the solo button is engaged (illuminated red), only channels that have been soloed will be mixed by the mixer. The other channels will effectively be muted. As with the mute button, the gain changes associated with soloing or unsoloing a channel are fully interpolated and will not cause audible clicks.

<Command>-clicking a solo button will exclusively solo the associated channel. The <command>-click will automatically clear the solo state of all other channels on the mix bus. <Option>-clicking a solo button will set all of the solos on the mix bus to the same state as the button you click. This allows you to quickly solo all the channels or quickly un-solo all the channels on a bus.

MIXER MASTER FADER

The Fader that appears on the right side of the mixer pane is the Mix Master Fader. This fader controls the overall bus level of the mix bus. Operationally, it is exactly the same as the channel faders (see: "**Channel Faders**" on page 37). Each Mix bus in the system has 24 dB of headroom above full scale. If the summing point of the mix bus is clipping, you can pull the mix out of clipping by dropping the Master fader, as long as the sum point is clipping by less than 24 dB.

MIXER MASTER MUTE

Each mix bus has a master mute button above the master fader that allows you to mute the output of that mixer. This mute is interpolated.

WIDE MIXERS

As was described before, the integrated mixers in Mobile I/O are **WIDE** – that is they allow you to mix every available input channel (both hardware channels and FireWire channels) together. In the widest case, the mixer will have 36 faders to allow you to control the gains for all of the input channels.

The MIO console window is only wide enough to accommodate 18 faders plus the Master fader. If you have enabled more channels than will fit in the window, the MIO console will automatically display a scrollbar at the bottom of the Mixer pane. Use this scrollbar to control which faders are visible at any given time. The width of the scrollbar indicator shows you how many of the enabled channels are visible at any given time. The scrollbar will automatically be hidden if you reduce the number of inputs to the mixer below 19 channels or if you switch to a different mixer that has less than 19 channels enabled.

TIP: Mobile I/O's support of near-zero latency mixing of every channel opens up a huge variety of applications that cannot be achieved with standard interfaces or, at the very least, require external gear or major work-arounds to accomplish. Some examples are:

1. Stem-based mixing.
   - In this mixing technique, you mix disparate elements of the program to separate sub-mixes called stems. You might mix drums to one stem, instruments to another, and vocals to a third. Then the relative balances of the mix can be addressed later in a macroscopic way (during mastering, for example). This also enables remixing the project easily without having to go back to the multi-track master. Since you will be creating stems on individual busses in the DAW, you need to sum the stems for monitoring. This is easily accomplished with Mobile I/O's mixer.

2. Monitoring mixed-in effects sends when the external effects are unavailable.
   - This is **Mobile** I/O, right? You may find that you want to continue editing or mixing while you are away from the studio. You can use the WIDE

mixer in Mobile I/O to mix in DAW effects send busses for monitoring without having to reconfigure your DAW session.

3. Multichannel foldback mixes.
   - Ever tried monitoring a full drum kit with a host-based foldback mix, or even a 2 channel cue mix? It can't be done! At least, it can't be done well). To monitor multichannel input tracking sessions you need an external mixer – or the built in WIDE-mixer in Mobile I/O.
   - If you take advantage of MIO's ADAT input with a third party A/D converter (or a second MIO) with ADAT outputs, you can simultaneously track and mix up to 18 input channels, with no latency, and no external mixer!

4. Near-Zero Latency monitoring of external effects.
   - Most singers need some reverb or other effects to get the feel right during their performance. With Mobile I/O and the WIDE mixer, you can split off a send from the performer's input channel, send it to an external effects unit, and mix the effect return into the performer's foldback mix -- with virtually no latency.

5. Multiple WIDE mixes
   - Since Mobile I/O supports multiple WIDE mix busses simultaneously, you can form multiple, individual foldback mixes for multiple performers at the same time, and best of all, each mix has its own complete mixer control surface, so you don't have to mess around with a million unreadable aux send knobs.

MIX/OUPUT ROUTING PANEL

The third panel in the MIO Console window is the **Mix/Output Routing** panel. This panel is what you use to access the powerful routing features that you won't find anywhere else. The Routing Matrix portion of the panel lets you dynamically control the configuration of the WIDE mixing engine and also configure the channel names for the hardware. You access this panel by clicking on the "**MIX/Output Routing**" button of the **view panel selector bar** at the top of the console window.
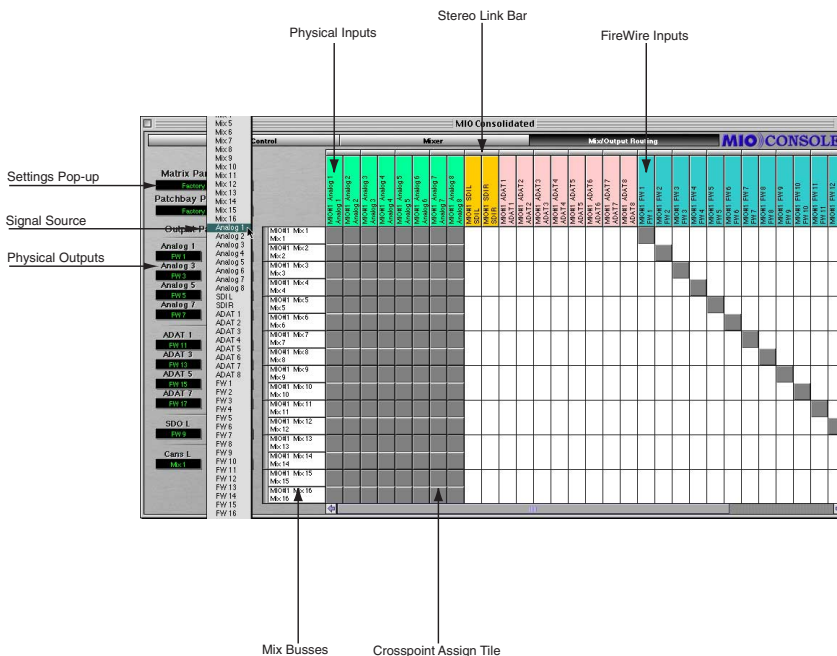
Figure 35: Mix/Output Routing Panel

The Mix/Output Routing Panel has two main components:

1. The Output Patchbay on the left side of the panel.
2. The Routing Matrix on the right side of the panel.

THE ROUTING MODEL

The Routing Panel provides a user interface for controlling the underlying routing architecture of the Mobile I/O. The routing architecture provides a

powerful routing model to allow you to control the routing of signals between physical & virtual inputs and the hardware mixer & physical outputs.

Conceptually, the architecture is quite simple:
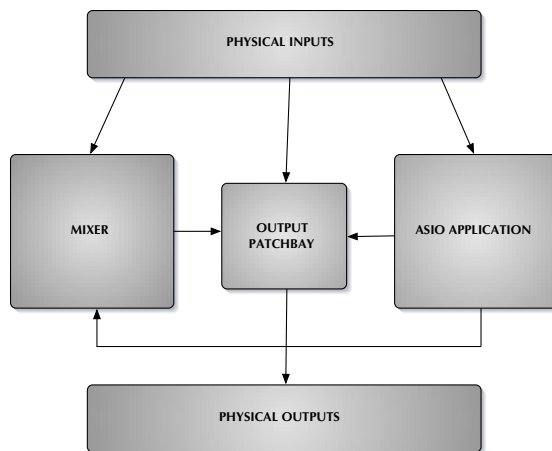
Figure 36: Block diagram of the Mobile I/O routing architecture

All of the physical inputs (e.g. Analog, ADAT and Digital), all of the channels being transmitted over the FireWire bus from the ASIO Application (e.g. DAW), and all of the outputs from the Mobile I/O WIDE Mixer are available to the Output Patchbay. The Output Patchbay can cross-point assign any of its inputs to any of the physical outputs (e.g. Analog, ADAT and Digital).

All of the physical inputs and all of the channels being transmitted over the FireWire bus from the ASIO Application are also inputs to the Mobile I/O WIDE Mixer. Every bus has each of those inputs available for mixing. The mixer outputs are sent to the Output Patchbay for routing to physical outputs. The number of mix busses varies with the Mobile I/O hardware model and the sample rate. The Mobile I/O 2882 supports 10 mono (5 stereo) mix busses at sample rates up to 50kHz (e.g. 1x rates) and 4 mono (2 stereo) mix busses at sample rates up to 96kHz (e.g. 2x rates).

Every physical input is mult'ed from the router/mixer section and sent directly over the FireWire bus to the ASIO host. Regardless of any mixing,

routing, or mult'ing that you configure in the hardware, you can always record all of the inputs with your DAW.

As you can see from this simple, high level view, the Mobile I/O routing architecture supports direct routing of any input to any output and also mixing of any set of inputs to multiple mixers. The outputs of the mixers can be routed to any output or any set of outputs for hardware mults. All hardware inputs are available to the ASIO host.

As it turns out, the Mixer/Router in Mobile I/O is quite extensive and the simple user interface hides a lot of complexity:
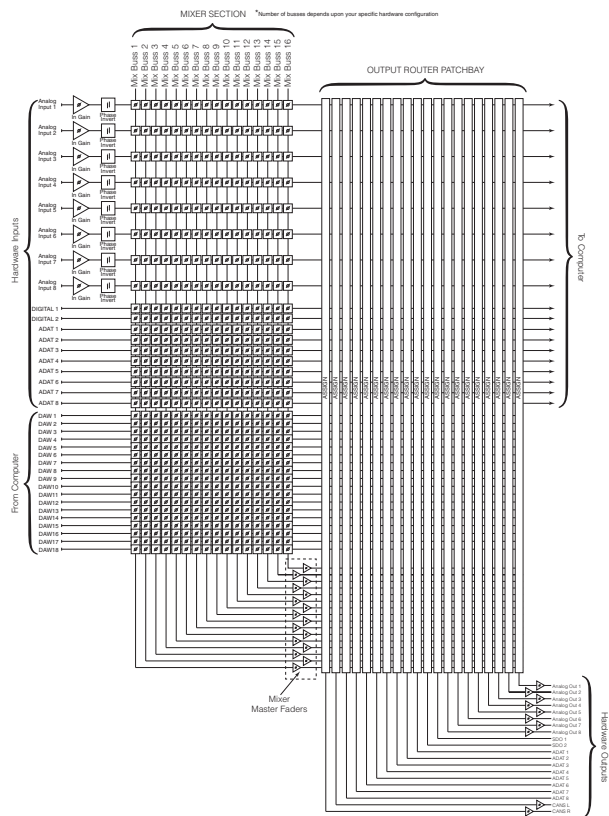
Figure 37: Complete view of the Mobile I/O Matrix Mixer

As you can see from this schematic diagram of the mixer/router, the Mixer in Mobile I/O is a fully cross-pointed matrix. While we are presenting the mixer to you as a set of mono and stereo mixers, the actual underlying structure is a full matrix. This means that the mixer is ready for surround and multi-channel matrix mixing today. The schematic provides an accurate representation that shows the signal flow through the mixer and into the router with all analog and digital gain points represented.

OUTPUT PATCHBAY DETAILS

The Output Patchbay allows you choose the signal sources to send to the hardware outputs. You can feed any output from your choice of a mix bus, a hardware input, or a FireWire input from the computer.

PATCHBAY POPUP CONTROLS

The output patchbay is the set of popup menus on the left side of the panel. Each popup menu corresponds to the hardware output that is labelling it. The channel that you choose from the popup menu will drive the associated hardware output.

Since the output patchbay is fully cross-pointed, you can easily construct direct routes from hardware inputs to hardware outputs for stand-alone converter operation, routes from ASIO (using the DAW channels) to the outputs for direct dubbing and monitoring from the computer, and routes from the mixer to hardware outputs for foldback, monitoring, and effects sends.

The output patchbay even lets you send the same channel to multiple outputs, which makes it really simple to create channel mults. You can use this to:

- send multiple copies of an aux group to different effects devices
- manage monitoring on multiple monitor systems (big and small)
- monitor on analog outputs and record on digital outputs
- send the same monitor mix to multiple outputs for multiple performers
- record safety mixes on digital two-track recorders while multitrack recording to the hard drive

To choose a source for an output channel, simply go to the popup menu for the output you want to route to, click and select the source send. Each popup window is like the jack of a patchbay, and each name in the popup menu is like a patchbay cable that carries the named signal.

PATCHBAY PARAMETER POPUP CONTROL

Directly above the patchbay popup menus is the "**Patchbay Parameters**" parameter popup control. This control allows you to store patchbay configurations for later recall. See the section "**Patchbay Popup Controls**" on page 45, for more details about the parameter popup menu.

The factory default routing for Mobile I/O configures the box to behave as a direct-routed audio interface – all the inputs go directly to the computer, all the outputs from the computer are directly routed to the hardware outputs, and the Headphone output is fed by the stereo mixer on Mix 1/2. This makes the Mobile I/O act like a basic audio interface.

Other configurations that may be very useful include:

- cross routing the analog inputs and outputs with the ADAT I/O – use this for 8x8 A/D/A operation (an upgrade for your ADAT or digital mixer!)
- direct routing of analog inputs to AES/SPDIF (2 channel Pre/DI/ADC)
- switching mults of the mixer outputs for alternate monitor configurations

This control allows you to maintain a library of frequently used routings and switch between them at will. Spend a little time to familiarize yourself with it – it is extremely powerful.

ROUTING MATRIX DETAILS

The **Routing Matrix** (Matrix) is the mixer assign matrix that is on the right side of the pane (see **Figure 35** on page 42). This control surface allows you to fully configure the structure of the multi-bus WIDE mixer in the Mobile I/O hardware. It also allows you to name all of the hardware and virtual channels that are accessible in the hardware.

You use the Matrix to assign hardware and FireWire inputs (DAW outputs) to mixers in the Mobile I/O. To assign channels to a given mix, you click on a

crosspoint in the Matrix, darkening the associated Crosspoint Assign tile. Deselecting a tile removes the associated fader from the mixer and mutes the channel in the hardware.

The WIDE mixer allows you to assign all of the hardware input channels and all of the FireWire channels to each mix bus. By limiting the number of channels assigned to any given bus, you can reduce the complexity of the associated mixer interface.

LOGICAL DESCRIPTION

Each mix bus and each input channel has a path label tile associated with it. These tiles are arrayed along the top and left edges of the Matrix.

The tiles along the top edge of the Matrix are the input path tiles. Each tile is color coded based upon the type of input it is:

- Green for Analog Inputs
- Orange for Digital Inputs
- Pink for ADAT Optical Inputs
- Blue for FireWire Inputs (Outputs from the DAW)

Each tile shows the Mobile I/O unit it is associated with, the physical name for the path, and the user defined name for the path.

Above each pair of input tiles is a thin tile that is used to join two input paths into a stereo input channel. In the factory default configuration, the Digital Inputs are joined into a stereo pair and each pair of DAW channels is joined into stereo pairs. If the channels are joined into a stereo pair and the pair is assigned to a stereo mixer, the pair will be represented by a single stereo fader in the mixer interface, and no pan knob will appear.

The white tiles along the left edge of the Matrix are the Mix bus tiles. All of these tiles are white because they all represent mix busses. Each tile shows the Mobile I/O unit it is associated with, the physical name for the mix bus, and the user name for the mix bus.

To the left of each pair of Mix tiles are thin inset tiles that are used to join two mix busses into a stereo bus. If the busses are joined, the associated mixer will have pan knobs for each of the mono hardware channels. If the

47

bus is a mono bus, there are no pan knobs associated with the input channels.

The interior of the Matrix is composed of a large number of square crosspoint assign tiles. The Matrix is too wide to fit completely within the MIO Console window; you may use the scrollbar that appears at the bottom of the Matrix to scroll the remaining DAW channels into view.

Each crosspoint assign tile indicates whether or not the input channel at the top of the column is assigned to the mix bus at the edge of the row. If the tile is filled in, the channel is assigned to the mix, and the controls for that channel will appear in the associated mixer. If the crosspoint tile is not filled in, the channel is not assigned to the mix, and the controls will not appear. The channel will be muted and un-soloed on the associated mix bus.

Stereo pairs are automatically assigned as a group to busses.

There are some keyboard shortcuts that you can use when making Matrix assignments:

- clicking in one tile, followed by <shift>-clicking in another tile will automatically select or deselect all of the tiles in between the endpoints; the selection will match the new state of the tile at the second click location
- clicking and dragging in the Matrix will continuously apply the new assignment state to each tile you drag over
- <Command>-click dragging will scroll the Matrix

CONFIGURING CHANNEL NAMES IN THE MATRIX
MIO Console has fully user configurable channel names. The names that you select for your channels will propagate to all of the other aspects of the MIO Console user interface. Due to limitations in the ASIO specification, the channel names will not propagate to your ASIO applications.

This allows you to name the channels in meaningful ways. The analog inputs can be named to match the sources. The Digital I/O can be named to match the effects device that you have patched. Mixes can be named by the foldback monitor or effects send that they will feed.

To name a channel, click on the input path tiles (for input channels) or the mix bus tile (for Mix busses). The channel configuration window will appear above the MIO console window:
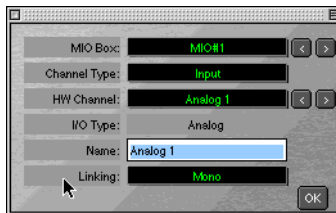
Figure 38: Channel Configuration Window.

All of the channel identification controls will be set for the tile you clicked on. To change the user selectable name of the channel, simply type the new name. The name will be updated in the console when you do one of the following things:

1. Hit the <return> key. This will update the name and dismiss the window.
2. Hit the <tab> key. This will update the name and switch to the next channel in the list.
3. Hit <shift><tab>. This will update the name and switch to the previous channel in the list.
4. Select any other channel using the popup menus or the "<" or ">" buttons in the channel configuration. This will update the name and switch to the new channel selection.

You can dismiss the window without updating the channel name by clicking its close box.

You can also select the stereo linking state from **Linking** popup menu in the channel configuration window. The state will be updated in the console along with the channel name.

The channel identification controls identify which channel you are adjusting:

- The **MIO Box** popup menu allows you to select which Mobile I/O you want to configure if more than one Mobile I/O is present on the bus.

- The **Channel Type** popup menu lets you switch between input channels and mix busses.

- The **HW Channel** popup menu identifies which hardware channel you are adjusting.

- The **<** button associated with the MIO Box and HW Channel popup menus steps to the previous item in the associated list.

- The **>** button associated with the MIO Box and HW Channel popup menus steps to the next item in the associated list.

MATRIX PARAMETER POPUP

The entire state of the Matrix can be saved to and recalled from the Console Parameter Library System. The basic functions of the Parameter Popup control are documented in the next section (see "**Parameter Popup Controls**" on page 50). Since you are very like to have a number of tracking and mixing configurations you use over and over again, the Parameter Popup for the Matrix is a real timesaver. This control appears in the top left corner of the pane, above the Parameter Popup control for the Patchbay router. Each time you create a configuration that you are likely to use again, save it in the Parameter Library for instant recall when you need it next.

PARAMETER POPUP CONTROLS

The **Parameter Popup** control is MIO Console's unified mechanism for handling presets for the various sections of the Mobile I/O. Each element of the console that supports the Parameter Library mechanism has a parameter popup control associated with it. These elements currently include:

- Input Channels

- Output Channels

- Mixers

- Output Patchbay

- Matrix

Each instance of the Parameter Popup control provides the same commands and options for every section of the console.

The parameter popup provides a hierarchical, categorized library of configuration presets for the associated section of the console. The menu is divided into three portions. The first portion consists of all of the items above the "Factory Default" item. The second portion is the "Factory Default" item and the third portion is the hierarchical items below the "Factory Default" item (see **Figure 39** on page 51).
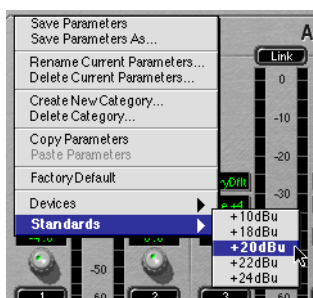
Figure 39: Parameter Popup Menu

The commands in the first portion of the menu allow you to save and manage the presets in the library. All of the presets are shared between like elements in the console. The preset commands are:

- Save Parameters

    use this command to save the current state of the associated console settings to the currently selected preset (will appear as "Save Parameters As…" if there is no currently selected preset)

- Save Parameters As…

    use this command to name and select a category to save the current state of the associated console settings as a preset into the library

- Rename Current Parameters…

    use this command to rename the currently selected preset

- Delete Current Parameters

  use this command to remove the currently selected preset from the library

- Create New Category

  use this command to add a new category to the library

- Delete Category

  use this command to delete the currently selected category and all of its associated presets

- Copy Parameters

  use this command to copy the current state of the associated console settings to the clipboard; you can use this to copy your settings from one block to another

- Paste Parameters

  If the clipboard contains compatible settings, this command will be available and will set the current state of the associated console settings to the settings on the clipboard. Use this with the "Copy Parameters" command to duplicate settings from one channel to another or from one mix to another

The "**Factory Default**" command will set the current state of the associated console settings to the default settings.

POPUP PRESETS

In the third part of the menu, each of the categories will be listed as a hierarchical menu title. Each of the presets for each category will be listed in the submenu under the category menu. The currently selected category and preset are drawn in bold, so you will know what is currently active.

Selecting a preset from the menu will make that preset active and will set the current state of the associated console settings to the values contained within the preset. The name of the currently selected preset will be drawn in the popup area in the console window to indicate which preset is active.

If you change the settings in the console, the name of the preset will be drawn in italics indicating that the current settings differ from the selected preset.

For the Input and Output channels, you can hold down the <option> key while selecting a preset to automatically apply the preset to all of the other input or output channels.

To access the parameter popup for the mixers, either click and hold the associated mixer tab or <control> click the associated mixer tab.

We have provided an initial set of presets for the various parameter libraries. The presets for the output channels are relatively complete and give you an idea of the power and flexibility of this approach to parameter management. We will be adding presets on a regular basis – check the website for new presets.

FINDER MANAGEMENT OF PRESETS

The presets in the console preset library are saved as standard files on your hard disk. This means that, in addition to using the commands in the parameter popup menu, you can also manage the presets directly from the Finder. The presets are all contained in the "Parameter Library" folder that is in the same folder as the MIO Console Application. The "Parameter Library" folder in the Console Application folder may be an alias to the real "Parameter Library" folder – so you can move the Parameter Library folder to another location and simply make an alias to the folder and place it with the console application.
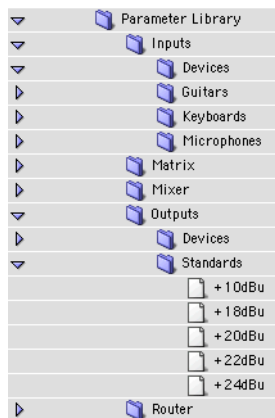

Figure 40: Parameter Libray Files in the Finder

You can create categories by creating new folders within the folders that appear within the "Parameter Library" folder. You can move presets from one category to another by moving the preset files within the hierarchy. You can also make copies by copying the files. You can delete presets by deleting the associated files.

Since the presets are stored as individual files, it is easy to share presets with other Mobile I/O users. By sending your preset files to another user, they can install your presets into their library by simply dropping the files into the appropriate folders in the Finder. You can rename categories by renaming the folders within the Finder.

Do **not** change the names of the "Parameter Library" folder or the top-level folders within the "Parameter Library" folder – these are used by the MIO Console library system to find the presets for the different sections of the console. The presets will not be found if those folder names have been changed (this includes the "Inputs," "Matrix," "Mixer," "Output," "Router," etc.)

You **cannot** rename presets from the Finder -- the parameter library will use the name stored in the preset file and ignore the file name in the Finder.

Any other changes you make in the Finder will not be recognized by the MIO Console while it is running – you will have to quit the Console and relaunch it for the changes to be recognized.

Routing Applications

Clocking Considerations

There are five ways that you can clock Mobile I/O:

1. Internal
2. Digital (AES/SPDIF)
3. Optical
4. Wordclock
5. 256X

Each choice is appropriate for a particular situation. In general, you should use Internal clock if you can because you are likely to find that it makes your other digital gear sound better. However there are some devices that either must be the clock master or work better if they are the clock master. For these devices, choose the clock port which is most appropriate for the device.

For example, many DAT machines distribute clock over their digital audio connection (AES or S/PDIF). For these machines you would connect to the AES or S/PDIF port on Mobile I/O and choose DigIn (44.1/48) for single rate or DigIn (88.2/96) for double rate in the MIO Console clock source popup. For more information about configuring the clock source, see "**System Controls**" on page 29.

SINGLE SPEED CLOCKING VS. DOUBLE SPEED CLOCKING

The difference between single speed (1x) and double speed (2x) clocking is handled as a fundamental mode change in Mobile I/O. The sample rate does not vary continuously between 48k and 88.2k but changes discontinuously when the double speed mode is set in the hardware.

When Mobile I/O is running on internal clock, this mode change is handled transparently when you specifiy what sample rate the box should use.

When Mobile I/O is running on external clock, it cannot determine in advance which mode to use, so you need to tell it what mode is appropriate for your external clock source. You communicate this information by select-

ing the proper external clock source. Each external clock source comes in two different flavors:

- **xxxx(44/48)**

- **xxxx(88/96)**

where the "xxxx" corresponds to the actual clock source. If you will be clocking off of a 1x source (e.g. 32kHz–50kHz sample rate) then choose the xxxx(44/48) flavor of the clock source. On the other hand, if you will be clocking off a 2x source (e.g. 64kHz–100kHz sample rate) then you will choose the xxxx(88/96) flavor of the clock source. If you choose the wrong flavor, Mobile I/O will not lock properly. When you select the flavor, it tells Mobile I/O which mode it has to run in, and the Mobile I/O can lock to the external clock in the appropriate frequency range.

EXAMPLE SETUP
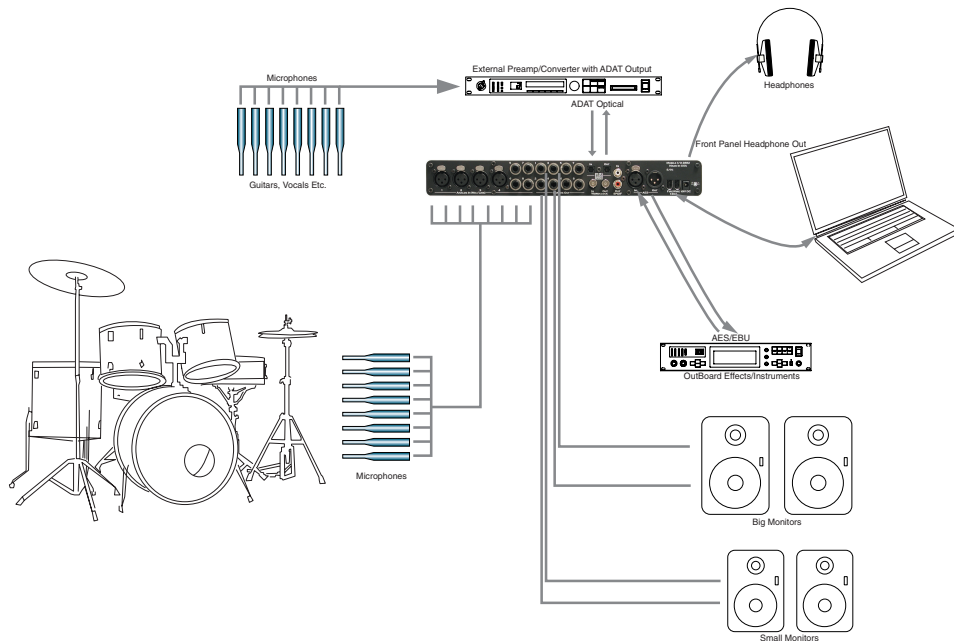
Figure 41: Example Mobile I/O Setup

In this section we will go step by step through some example applications that are possible with a Mobile I/O system configured as shown in Figure 41

on page 56. After reading this section you should have a feel for how to apply Mobile I/O's routing and mixing capabilities.

In this configuration Mobile I/O is connected to an external preamp/converter via ADAT Optical. This gives a total of 16 analog inputs. All of these are available for mixing with the WIDE mixer.

WHICH DEVICE IS THE CLOCK MASTER
The first consideration involved with a setup like this is which device should be the clock master. If the external device is happy slaving to Mobile I/O then make Mobile I/O the clock master by setting the clock source to internal in MIO Console. Set the external device to lock to ADAT Optical, or wordclock if the external device has a wordclock input. If you use wordclock, make sure to connect a 75 Ohm BNC cable between Mobile I/O and the external device.

If you find that you are getting clicks and pops in the audio coming from the external converter, try clocking in the other direction. Set the external converter to use its internal clock and set the Mobile I/O clock source to ADAT. Remember to choose the appropriate clock rate (single or double) for the sample rate you will be working at.

Once you have Mobile I/O and the external converter locked you should be able to get the AES effects device to lock to Mobile I/O's AES signal, regardless of whether Mobile I/O or the external converter is the clock master.

Now that all of the digital devices are playing nicely with each other, let's take a look at some possible applications with this setup.

EFFECTS FOR TRACKING
Many performers (especially singers) find it easier to get a good take if they have some sweetening effects in their headphone mix while tracking. These will usually be temporary effects and will not neccesarily make it into the final mix but they can make the difference between a good take and a bad take. In this example, we will be using an effects device connected to Mobile I/O's AES ports to add some reverb to the headphone mix we make in MIO Console.

The first thing we need to do is configure our mixers and routing. For this setup we will use two mixers in MIO Console: one for the actual headphone mix and one for the send mix to the reverb. To do this we need to bring up the Mix/Output Routing view in MIO console:

Figure 42: Configuring the Matrix
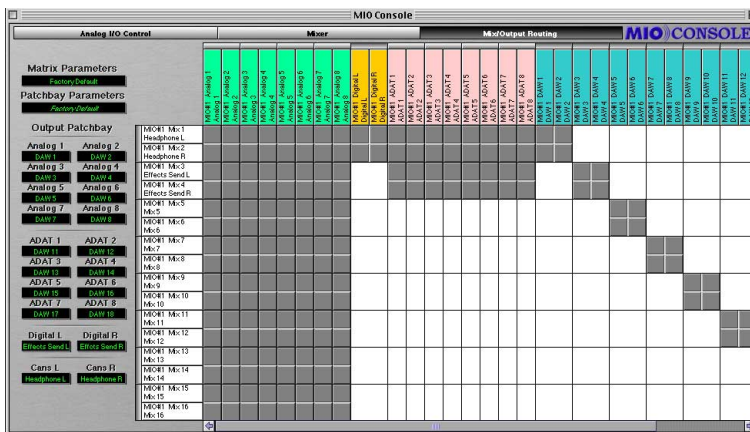
In its default state, the WIDE mixer is configured with only the Analog Inputs assigned to the mix busses, so we need to add the ADAT inputs. To do this we simply click in the assign tiles under the ADAT inputs for the mix busses we want. For this example we will use for the stereo headphone mix and mix busses 3 and 4 for the send mix.
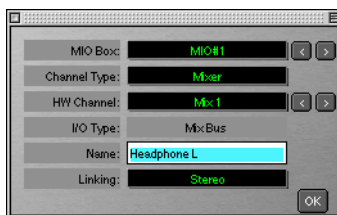

Figure 43: Naming the mix busses

We'll also assign DAW 1 and 2 to mix busses 1 and 2 for ASIO playback. We can name the mix busses by clicking on the mix bus label. This will show the matrix configuration window (Figure 43 on page 58).

We'll name the headphone mix Headphone L/R and the send mix Effects Send L/R. You must name each mono bus separately, and the Console combine the names automatically. You can use the tab key to increment through the mix busses as you name them. When you're finished, hit the return key to close the configuration window.

The mixer bus names have propagated through MIO Console which means they will now show up in the Output Patchbay and the Mixer view.

Note that Digital L/R is assigned to the Headphone mix but not the Effects send mix. This is critical because Digital L/R is the effects return, and we definitely do not want to assign the return to the send. If we do, we'll get feedback and no one, especially the singer, will be happy.

To route the mixers to their intended destinations we use the Output Patchbay. We want the Headphone mix to feed the front panel headphone output of Mobile I/O so we'll set the Cans section of the Output Patchbay to take its input from Headphone L/R.

Figure 44: Making the Output Assignment

The Effects Send mix should feed the AES outputs so we'll set the Digital section of the Output Patchbay to take its input from Effects Send L/R.

The routing is now set. We can save the setup for the next tracking session using the **Matrix Parameters** and **Patchbay Parameters** popup menus (see "**Parameter Popup Controls**" on page 50 for details).

Now we'll switch to the Mixer View.
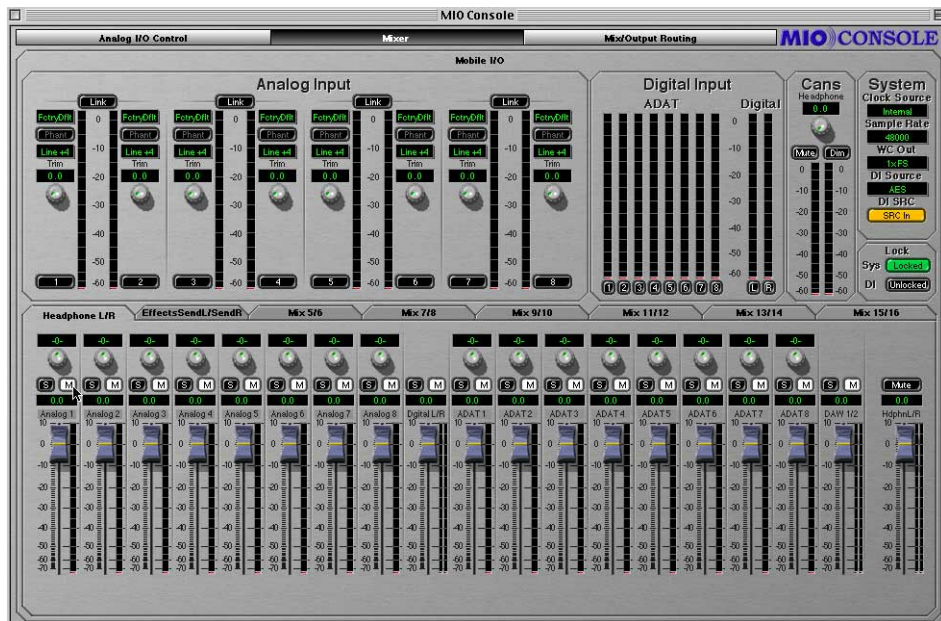

Figure 45: Headphone Mixer

Notice the Mix Tabs on the left are now labeled Headphone L/R and Effect send L/R. Also notice the fader marked Daw 1/2 on the right. We'll use this fader to control the playback level of the ASIO tracks we want to overdub against.

If you have signals running to Mobile I/O you should see activity on both the input meters and the mixer meters.

By default the mixer's faders are are at unity gain and the mutes are engaged. We can unmute all the mixer channels by holding the option key and clicking one of the mutes. All of the channels will unmute. But for the time being we'll keep the Digital L/R fader muted.

Now we can set up a dry headphone mix. Once we have basic balances set, we can switch to the Effects Send mixer by clicking its mix tab. In the Effects Send mixer we can unmute each channel and set its send level to the reverb, or we can use the Mixer parameters popup (see "**Mixer Pane Tabs**" on page 36) to copy the mix from the Headphone mixer to use as a starting point.

To copy mix parameters from one mixer to another:

1. Click and hold the mix tab for the mixer you want to copy from. A popup menu will appear:

Figure 46: Mixer parameters popup

2. Choose Copy Parameters
3. Now click and hold on the mix tab you want to copy to
4. Choose paste parameters
5. The mix parameters have now been copied

Now we can can go back to the Headphone mixer and unmute the Digital fader. The reverb should now be heard in the headphone mix.

If you want to use this example as a tracking set up, choose **Save As...** from the file menu. This brings up a standard Macintosh save dialog and allows you to save the entire state of MIO Console as a setup document.

CONTROLLING MULTIPLE MONITORS

We can use the routing snapshot features of Mobile I/O to set up multiple monitor paths and easily switch between them.

For this example we will have two sets of powered monitors connected to Mobile I/O. The first set will be connected to Analog Outputs 1-2 and the second to Analog Outputs 3-4.

In the routing window we'll set up a monitor mixer on busses 5 and 6 and name it Monitor L/R. We'll assign Analog 1-8, ADAT 1-8, Digital L/R and DAW 1/2 to the monitor mixer. This will allow us to monitor playback from an ASIO application as well as our live inputs.
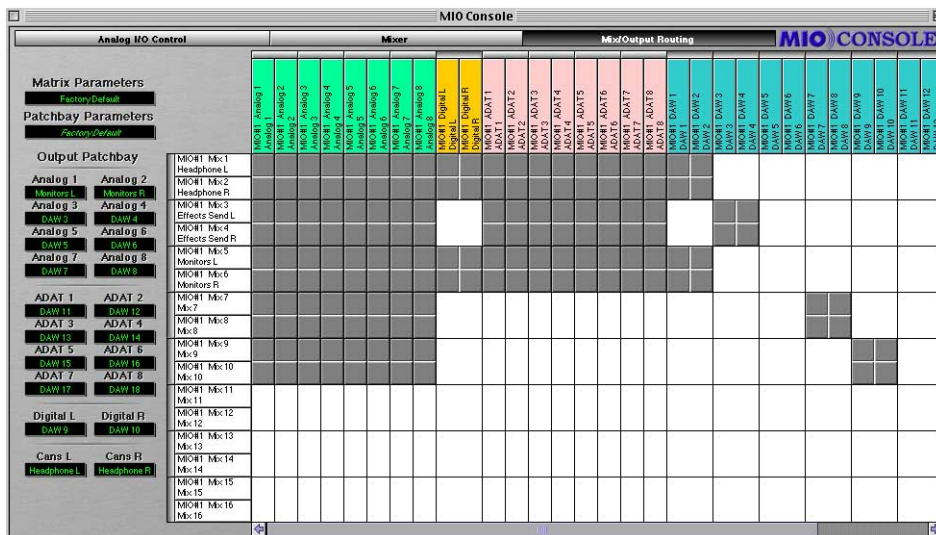
Figure 47: Setting up the monitor mixer

Now in the Output Patchbay we'll assign Monitor L and Monitor R to Analog 1 and Analog 2. This creates the routing for our Big Monitors. We'll also make sure to assign an output pair that we know won't ever have signal on it to Analog 3 and Analog 4. We'll use Daw 17 and Daw 18.
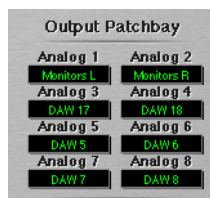
Figure 48: Setting the Monitor path

We want to make sure that Analog 3 and Analog 4 won't ever have signal in this setup because the Small Monitors are connected to Analog 3 and Analog 4 and we don't want to have random channels playing through the Small monitors while we are listening to the Big Monitors.

Next we'll save this routing using the Patchbay parameters popup menu. But first we'll create a new category by choosing **Create new category...** from the Patchbay Parameters popup.

A dialog will appear:
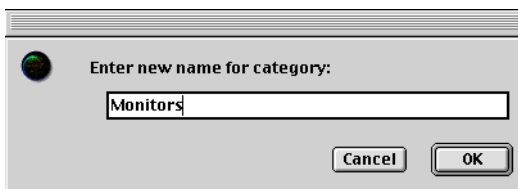
Figure 49: Naming a category

We'll name the category "Monitors" and click **OK.**

63

Now we can save the setup into the Monitors category by choosing **Save As...** from the Patchbay Parameters popup. A dialog will appear:
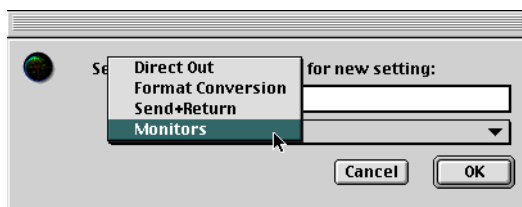
Figure 50: Saving a setting

We'll choose the Monitors category from the category popup, name the setup "Big Monitors" and click OK. The setup has now been saved.

Now we can set up the routing for the Small monitors. To do this we simply assign the monitor mixer to Analog 3 and Analog 4 and assign DAW 17 and DAW 18 to Analog 1 and Analog 2.
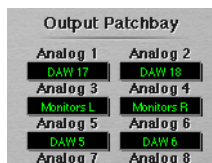
Figure 51: Small Mixer setup

We'll save the setup into the Monitors category as "Small Monitors" using the Parameter Patchbay popup. Now we have a monitor switching system. To switch monitors, simply choose the monitor path you want using the patchbay popup window.
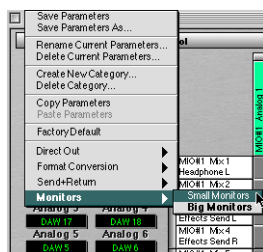
Figure 52: Choosing the Monitor Path

Conclusion

The preceding examples should give you a sense of the possibilities that are enabled by the routing and mixing features of Mobile I/O. While this is just a starting point, we have covered all of the basic operations required to manipulate Mobile I/O with complex routing. You should be able to build upon these scenarios to construct routings that suit your needs and your workflow.

We will be publishing a series of application notes on the

http://www.mhlabs.com/mio

website that will explore other, more complex routings for controlling common types of recording and monitoring setups – all from Mobile I/O. Please check in from time to time to see what's new.

Configuring ASIO Hosts

ABOUT ASIO™ TECHNOLOGY

ASIO (Audio Stream Input/Output) is Steinberg Soft- und Hardware GmbH's technology standard for interfacing professional audio applications to professional audio interface hardware. Steinberg defined the standard and used it for all of their audio applications, and took the further step of making the specification of the standard available for use by other hardware and software vendors. In doing so, they established it as the de facto standard for making high-resolution, multi-channel, low-latency connections between audio hardware and audio applications on Mac OS 9 and Microsoft Windows.

Virtually every pro audio computer application provides support for communicating with audio hardware via ASIO. As such, it was the natural standard for Metric Halo to support for interfacing with Mobile I/O. Applications (programs) that communicate with hardware via ASIO drivers are called ASIO Hosts.

The ASIO standard is quite rich and provides many possible options for variations in the specification. There are many aspects of the spec that are not supported by all ASIO hosts (e.g. ASIO direct monitoring).

Metric Halo has done extensive testing with the major ASIO hosts, and some of the minor ones, and has worked to ensure maximum compatibility with all hosts. Even if you are using a host that is not specifically discussed here, you are unlikely to encounter problems. If you do, please file a bug report with both Metric Halo and the developer of the ASIO host.

ASIO Basics

HOW THE ASIO DRIVER WORKS

The Mobile I/O driver is broken into two components. The first component is composed of the "**MobileIO Driver**" and "**MobileIO Enabler**" files, which are installed in the **Extensions** folder in the **System Folder**, and are the core of the Mobile I/O driver (the enabler file allows the Mac to recognize the Mobile I/O). The **MobileIO Driver** file is responsible for connecting the computer to the Mobile I/O hardware, managing the driver state, facilitating the

communication between MIO Console and the hardware, and transporting audio between the hardware and clients. The "**Mobile IO™ ASIO**" file, which is installed in the "**ASIO Drivers**" folder of each ASIO host, is a client of the **MobileIO Driver**, and provides a gateway between the ASIO host and the Mobile I/O driver. This allows us to change and enhance the **MobileIO Driver** without having to have third parties update their applications to support new functionality in Mobile I/O – as long as the functionality can be mapped onto the commands specified by ASIO. With the current version of the Mobile I/O driver, only one ASIO client can be attached to the driver at any one time. This may change in the future.

WHERE THE ASIO DRIVER FILE GOES

Most ASIO hosts use an "**ASIO Drivers**" folder located in the same folder as the host application. If your host does not use the standard mechanism for organizing and locating ASIO drivers, you will have to refer to the documentation that accompanies your host for information about the appropriate location for the **Mobile IO™ ASIO** driver file.

ASIO TRANSPORT AND SAMPLE RATES

The ASIO specification supports a wide variety of audio transport standards, but most applications only implement a small subset of the possibilities. As a practical matter, ASIO supports multichannel transport of 24 bit audio at virtually any sample rate. Many hosts only support a small number of sample rates and may not support all of the sample rates that are available with the Mobile I/O hardware.

CHANNEL NAMES

ASIO provides a mechanism for the driver to tell the host the names of the channels. This mechanism is not dynamic, so the Mobile I/O driver cannot update the host's names as you adjust the Output Patchbay router. As a result, the input channel names are always accurate, but the mapping between the ASIO output names and the hardware outputs can be adjusted on the fly, and the host may not match the actual hardware routing.

The Output Channel names reported to the host are the default direct mapped hardware outputs. This is the default configuration when the box

boots up. In the direct mapped configuration, the 18 ASIO output channels are mapped to the hardware in the following order:

1. DAW 1 $\Rightarrow$ Analog 1
2. DAW 2 $\Rightarrow$ Analog 2
3. DAW 3 $\Rightarrow$ Analog 3
4. DAW 4 $\Rightarrow$ Analog 4
5. DAW 5 $\Rightarrow$ Analog 5
6. DAW 6 $\Rightarrow$ Analog 6
7. DAW 7 $\Rightarrow$ Analog 7
8. DAW 8 $\Rightarrow$ Analog 8
9. DAW 9 $\Rightarrow$ Digital 1
10. DAW 10 $\Rightarrow$ Digital 2
11. DAW 11 $\Rightarrow$ ADAT 1
12. DAW 12 $\Rightarrow$ ADAT 2
13. DAW 13 $\Rightarrow$ ADAT 3
14. DAW 14 $\Rightarrow$ ADAT 4
15. DAW 15 $\Rightarrow$ ADAT 5
16. DAW 16 $\Rightarrow$ ADAT 6
17. DAW 17 $\Rightarrow$ ADAT 7
18. DAW 18 $\Rightarrow$ ADAT 8

You can use the table above as a guide to the mapping between ASIO outputs and DAW FireWire channels when you want to adjust the direct routings in the Output Patchbay Router.

CHANNEL ENABLES

ASIO supports enabling and disabling channels. Some hosts do this automatically (e.g. Digital Performer), some hosts provide manual controls (Cubase, Nuendo), and other hosts do not provide explicit control for enabling and disabling channels. In order to reduce CPU load, Mobile I/O's ASIO driver takes notice of disabled channels by skipping processing for those channels. For applications that provide manual control over enabled channels, you will get the best performance if you only enable the channels you need. For applications that automatically maintain the enabled channels, you will get the best performance if you only assign channels to outputs you really want to use, and only record enable channels that you intend to record on.

ASIO BUFFERS

Audio channels are transported individually in buffer sized chunks. The size of the audio buffers has an effect on the CPU load of the audio application, as well as the round-trip latency from input to output when the audio is routed through the host application for monitoring or processing.

Generally, the CPU load increases as the buffer size decreases (this is not true for all hosts; emagic's Logic Audio is a notable exception). On the other hand, the latency decreases as the buffer size decreases.

Since, in general, you want the lowest CPU load **and** the lowest latency, you will have to make trade-offs.

The WIDE mixer engine in Mobile I/O helps substantially with this issue, because for the common critical monitoring configuration (monitoring while tracking external sources), the Mobile I/O mixer removes all of the ASIO latency from the monitor path and allows you to decouple the latency from the buffer size.

In the case that you are trying to perform with a softsynth running in your host, Mobile I/O's mixer does not help decrease the latency since the signal is being generated on the computer. In this case you'll want to minimize the output latency by selecting the smallest ASIO buffer size possible. This will depend on your computer hardware, the amount of processing you are doing, and the ASIO host you use.

SETTING THE ASIO BUFFER SIZE

The methods used to set the ASIO buffer size will vary from host to host. Some ASIO hosts provide direct controls for adjusting the buffer size and others do not. If your host supports setting the buffer size directly, you should take advantage of that facility. Most hosts that support controlling the buffer size will automatically save your selected buffer size with your session.

Mobile I/O supports buffer sizes that are powers-of-two. This means that you can use buffers that are 64, 128, 256, 512, 1024, or 2048 samples long. Some hosts that allow you to change the buffer size do not properly follow the ASIO spec and will let you specify buffer sizes that are not supported by

the driver. If your host allows you to select a buffer size, please make sure that you choose one of the sizes that Mobile I/O supports (Digital Performer is an example of a host that lets you set the size, but lets you set it to values that are not valid).

If your host does not support setting the buffer size, it will generally use the buffer size that the Mobile I/O driver recommends. The default buffer size for Mobile I/O is 1024 samples per buffer.

The "Driver Options" dialog supported by most ASIO hosts provides a popup menu to allow you to set the default buffer size reported by the Mobile I/O driver to the ASIO host. Most hosts will take notice of a new setting immediately, but some will not. If your host does not appear to notice that the buffer size changed (e.g. the latency does not change), you will have to select another ASIO driver and then switch back to the Mobile I/O driver for the changes to take effect (Cubase VST is an example of a host that will not notice the change automatically).

SAMPLE SIZE

The Mobile I/O ASIO driver provides the ASIO host with 24-bit samples. If you record with a 16-bit session, there is no mechanism for the ASIO host to inform the Mobile I/O driver that it wants 16-bit samples. It is the responsibility of the host to dither the incoming audio to 16-bit samples before recording them. If the host does not dither the samples to 16-bit, they will be truncated by the host when they are recorded. For best recording quality, use your host's 24-bit recording option.

CLOCK SOURCES

The ASIO specification provides the capability for hosts to control the hardware clock source. Some hosts provide a user interface to do this, others do not. If the host does provide an interface to do this, you will be able to select one of the Mobile I/O external clock sources directly from the host. If the host does not provide an interface, you will need to use the MIO Console to select the external clock source. If you have selected an external clock source using either the host or the console, you will not be able to control the system sample rate from the computer. MIO Console will automatically reflect the clock source and sample rate set by the host.

ASIO DIRECT MONITORING

The ASIO2 standard added the capability for ASIO hosts to control hardware monitor mixing in ASIO hardware. The Mobile I/O ASIO driver maps ASIO Direct Monitoring commands to the Mobile I/O WIDE hardware mixer. The current implementation of the driver does not share mix configuration data with MIO Console, so if you enable ASIO Direct Monitoring, you will be able to change the state of the MIO Mixer without the changes being reflected in the MIO Console user interface.

For basic multitrack foldback, MIO Console provides detailed control and metering and may provide a more streamlined interface for controlling foldback mixes.

The exciting part of ASIO direct monitoring is that it enables "tape-type" cross-faded punch-ins when doing overdubs. Hosts that support tape-type monitoring and ASIO Direct Monitoring will automatically mute the input channel in the monitor mixer when the transport is in playback, and then will unmute the monitor channel when the engineer punches in. The Mobile I/O's interpolated mixer provides clickless, cross-faded transitions for punch-ins and punch-outs, allowing you to accomplish clean, transparent overdubs.

In order to take advantage of the direct monitoring features, you will need to configure both your ASIO host and your Mobile I/O:

1. Enable ASIO Direct Monitoring in your host. Refer to the documentation that comes with your host for details. Later in this chapter we provide details about the ASIO implementation of some of the primary ASIO hosts. That documentation includes details about ASIO Direct Monitoring.

2. Use the Output Patchbay (see "**Output Patchbay Details**" on page 45) in MIO Console to route the appropriate Mix Busses to your monitor output path (e.g. Analog 1/2 or Cans L/R).

   • The ASIO Direct Monitor will assign your monitored input channels to the mixer that matches the output bus that the input channels are assigned to in the Host.

   • If the input channels are being mixed onto Analog 3/4, the ASIO Direct Monitor will assign the input channels to Mix 3/4 – your mixes must match your outputs for the Direct Monitor to work.

3. Enable the appropriate Monitoring mode in your host software if it supports multiple monitor modes.

4. Make sure that the mixer does not have solo'ed channels and that the Mixer master mute is not enabled – if either of these gotchas are in force, you will encounter problems controlling the mixer from ASIO.

CONFIGURING NUENDO

Nuendo is a Steinberg application, and it uses ASIO as its only audio interface standard. The ASIO driver is selected from the VST Multitrack section of the Device Setup dialog:
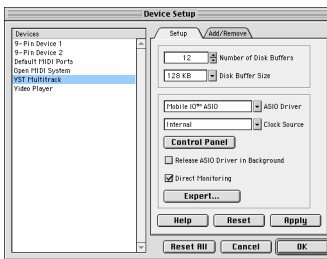

Figure 53: Nuendo VST Device Setup Dialog

If you want to enable ASIO direct monitoring for Nuendo, you enable the "Direct Monitoring" checkbox in the Device Setup dialog.

The "Control Panel" button accesses the Mobile I/O driver options dialog, which allows you to set the ASIO buffer size. Nuendo applies the new buffer size when you click the Apply button or the OK button.

There are some advanced settings in the Nuendo Device Setup Dialog that you need to select for proper operation. You can access these settings by clicking the "Expert…" button. The VST Engine Expert Settings Dialog will appear:

Figure 54: Nuendo VST Expert Options

You should make sure that the Lower Latency and Compatibility Mode checkboxes are enabled – if they are not enabled, you will encounter problems. The other elements of the expert options are not critical, but you may want to use the settings we show here as they have been tested extensively with Mobile I/O.

The Device Setup Dialog also provides access to Mobile I/O's external clock sources from the "Clock Source" popup menu. This menu contains the same options that appear in the MIO Console clock source popup menu.
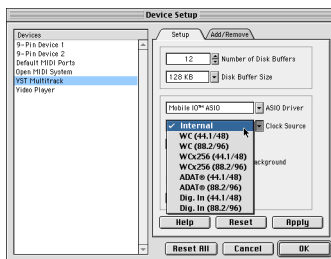
Figure 55: Nuendo Clock Source Popup

Nuendo's VST Inputs window provides access to the controls to name the ASIO input channels. It also allows you to enable input channels. The port names that appear on the left hand side of the window are the physical input ports reported to Nuendo from the driver. The names on the right side of the window are user configurable. The "On" buttons down the middle of the window control which channel pairs have been enabled in the driver. You need to enable these to assign input strips to the hardware channels, but

73

under normal circumstances, you should only enable the channels you need. Especially if you are mixing and pushing your processor to the limit, you can gain some extra processor power by enabling the minimum number of input channels.

Figure 56: Nuendo VST Inputs

The VST Outputs window allows you to enable which physical outputs you want to mix to. As with the inputs, you can save some processing power by using the minimum outputs required to do the job.

Figure 57: Nuendo VST Outputs

TIP: Enabling and disabling an output bus causes Nuendo to reset the driver. If you encounter any driver synchronization problems, this is the fastest way to reset the driver and force a resync.

CONFIGURING CUBASE

Cubase is a Steinberg application that uses ASIO as its only audio interface standard. The ASIO driver is selected from the "ASIO Device" popup menu of the "Audio System Setup" dialog:
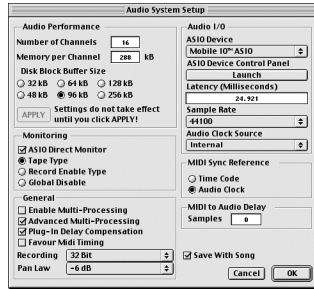

Figure 58: Cubase System Setup

If you want to enable ASIO direct monitoring for Cubase, enable the "ASIO Direct Monitor" checkbox in the Audio System Setup dialog.

The "Launch" button accesses the Mobile I/O driver options dialog, which allows you to set the ASIO buffer size. Cubase does not apply the new buffer size automatically. You will need to switch the ASIO Device popup to another driver and then back to Mobile I/O for the buffer size change to take effect.

The Audio System Setup dialog also provides access to the sample rate and clock source settings for the Mobile I/O driver.

Cubase's VST Inputs window provides access to the controls to name the ASIO input channels. It also allows you to enable input channels. The port names that appear on the left hand side of the window are the physical input ports reported to Cubase from the driver. The names on the right side of the window are configurable by you. The "Active" buttons down the middle of the window control which channel pairs have been enabled in the driver. You need to enable these to assign input strips to the hardware channels, but under normal circumstances, you should only enable the channels you need. Especially if you are mixing and pushing your processor to the limit, you can gain some extra processor power by enabling the minimum number of input channels.

Figure 59: Cubase VST Inputs

The VST Outputs window allows you to enable which physical outputs you want to mix to. As with the inputs, you can save some processing power by using the minimum outputs required to do the job

TIP: Enabling and disabling an output bus causes Cubase to reset the driver. If you encounter any driver synchronization problems, this is the fastest way to reset the driver and force a resync.

CONFIGURING LOGIC

Logic Audio supports multiple driver types in addition to ASIO. In fact, Logic supports using multiple driver types at the same time. Refer to your Logic documentation for information on how take advantage of this feature.

Configuring Logic to use Mobile I/O as one of its I/O devices is done from the Audio Driver section of the Logic Preferences. This section of the preferences window displays a checkbox for each of the supported driver types. You must enable the ASIO driver type by checking the ASIO checkbox. Make sure that the other options for the ASIO driver are displayed as shown in *Figure 60 on page 77*.

To select Mobile I/O as your ASIO driver, choose "Mobile I/O ASIO" from the Driver popup menu.

To adjust the buffer size, click the "Control Panel" button to access the Mobile I/O driver preferences. Logic does not automatically reset the buffer size. In order to have the change take effect, you need to reset the driver. You can do this by cycling the state of the "Larger Process Buffer" checkbox.

You can choose an external clock source from the "Clock Source" popup menu.
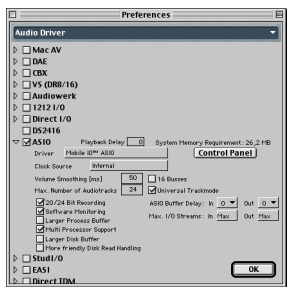

Figure 60: Audio Driver section of the Logic Preferences

Finally, to enable ASIO Direct Monitoring, you must **uncheck** the "Software Monitoring" check box in the Audio Driver preferences.

CONFIGURING DIGITAL PERFORMER
Digital Performer supports multiple driver types including ASIO through the MOTU Audio System (MAS). MOTU has provided an ASIO wrapper for MAS that allows MAS to use ASIO drivers as if they were built into the MAS system, albeit with some extra overhead. Unlike the other ASIO Hosts described in this chapter, MAS appears to have significant sensitivity to ASIO buffer sizes – running MAS with 128 sample buffers will double the CPU load over 1024 sample buffers. As a result, you have to carefully trade-off the total amount of processing against output latency when you work with Digital Performer.

SELECTING MOBILE I/O
Configuring MAS to use Mobile I/O as its I/O device is done from the **Configure Hardware Device** dialog of Digital Performer. In order to select Mobile I/O as your interface, you must first select ASIO from the popup menu at the top of the window (see *Figure 61 on page 78*). After you have selected ASIO, you will be able to choose **Mobile I/O ASIO** from the ASIO Driver popup menu. MAS may warn you that the driver has not been tested and may cause problems. **You may safely ignore this warning**.

The **Configure Hardware Device** dialog provides a user interface to configure the clock source, sample rate, and buffer size for the Mobile I/O ASIO driver. All of these settings are saved with your session and therefore are the appropriate mechanism for controlling the driver settings.
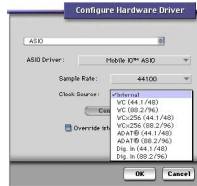
Figure 61: DP/MAS Configure Hardware Driver dialog box

If you select an external clock source from the Clock Source popup menu, the external source will determine the sample rate and you will not be able to control the rate from the computer.

Clicking the **Control Panel** button will bring up the Mobile I/O Driver preferences dialog. In general, you do not need to use this dialog with MAS since MAS provides a user interface for controlling the buffer size.

The "**Override Buffer Size**" checkbox enables MAS to set the buffer size. If the checkbox is unchecked (unfilled), MAS will use the driver specified default buffer size. If you check the "**Override Buffer Size**" checkbox MAS will set the ASIO buffer size to number in the text box to the right of the checkbox. MAS allows you to set that number to any arbitrary value. The MAS/Mobile I/O Driver combination will not work correctly unless you specify one of the valid buffer sizes (64, 128, 256, 512, 1024 or 2048).

WARNING: Since the buffer size is saved with the session in Digital Performer, and using smaller buffer sizes increases the CPU load, you may encounter problems opening heavily loaded sessions on slower machines. If you overload a machine due to having saved a small buffer size with a session, MAS will warn you that the CPU has overloaded and will allow you to disable your MAS plug-ins. If this happens, open the **Configure Hardware Device** dialog and increase the buffer size. When you dismiss the **Configure Hardware Device** dialog, the plug-ins will be re-enabled.

ENABLING CHANNELS

MAS automatically enables and disables ASIO channels based upon the assigned output busses and the record-enabled channels and Aux inputs. You can use this fact to conserve CPU power by only record-enabling channels when you intend to record on them, only turning on hardware aux inputs when you are using them, and only mixing to hardware busses when they are being used.

ASIO DIRECT MONITORING

Digital Performer does not support ASIO Direct Monitoring.

CONFIGURING SPECTRAFOO

In order to use ASIO drivers with SpectraFoo or SpectraFoo Complete, you must have the Rad3f18 release or newer. The f18 release is a free upgrade available from Metric Halo.

SpectraFoo is a multi-driver application that supports all ASIO devices using SpectraFoo ASIO support. To choose an ASIO driver, you select ASIO from the SpectraFoo "**Audio I/O**" menu.
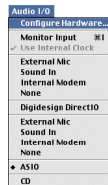

Figure 62: SpectraFoo Audio I/O menu

After ASIO has been selected, you configure the ASIO driver from the "**ASIO Driver Configuration**" dialog. Select the the "**Configure Hardware…**" command from the "**Audio I/O**" menu to show the "**ASIO Driver Configuration**" dialog:
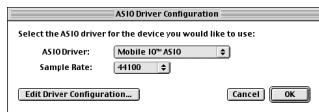

Figure 63: SpectraFoo ASIO Driver Configuration dialog

79

To use Mobile I/O with SpectraFoo, select "**Mobile IO™ ASIO**" from the ASIO driver popup menu. After you have selected the driver, you can set the Sample Rate from the Sample Rate popup menu. SpectraFoo supports all of Mobile I/O's sample rates.

Use the "**Edit Driver Configuration…**" button to adjust the ASIO buffer size. SpectraFoo adjusts the buffer size as soon as you change it.

SpectraFoo currently uses all the available hardware channels for instant matrixing, so there is no way to enable or disable ASIO channels.

Troubleshooting Guide

COMPUTER DOES NOT SEE MOBILE I/O

If you attach Mobile I/O to your computer, and the computer is unable to communicate with the Mobile I/O hardware there are five basic possibilities for the source of the problem:

1. The Mobile I/O is not powered up
2. The Software is not installed properly
3. The FireWire bus did not reset correctly
4. The FireWire cable is bad
5. The FireWire hardware has been damaged

MOBILE I/O IS NOT POWERED UP

The first thing to check is that the Mobile I/O is, in fact, powered up.

If Mobile I/O is powered up and booted properly, the Power, Sample Rate, and Locked front panel indicators will be illuminated. If these indicators are not illuminated, the Mobile I/O is not powered properly or the unit's firmware has been corrupted. If you determine that you are powering Mobile I/O properly and the indicators are not illuminated, you will need to contact Metric Halo support.

If you are bus powering the Mobile I/O, there is a possibility that you have overloaded the power rating of the power source. Please see the troubleshooting section "**Not enough power on the bus**" on page 90 for details on troubleshooting this problem.

If the Mobile I/O is properly powered, then check the next possibility.

SOFTWARE IS NOT INSTALLED PROPERLY

In order for the computer to properly communicate with the Mobile I/O the various components of the driver software need to be installed correctly. If the software is not installed correctly, the communication between the computer and Mobile I/O will fail in various ways:

1. If the "MobileIO™ Enabler" file is not installed in the "System Folder:Extensions" folder, the system will not register the attachment of a Mobile I/O device to the bus.

   • The symptom of this is that the Front Panel FireWire indicator is illuminated, but the ASIO driver reports that the Mobile I/O is not present and
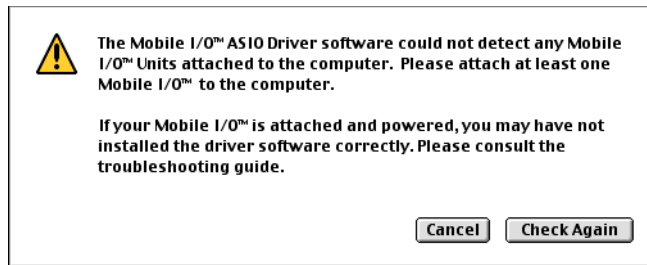

The Mobile I/O™ ASIO Driver software could not detect any Mobile I/O™ Units attached to the computer. Please attach at least one Mobile I/O™ to the computer.

If your Mobile I/O™ is attached and powered, you may have not installed the driver software correctly. Please consult the troubleshooting guide.

[Cancel] [Check Again]

Figure 64: Mobile I/O not found dialog

   the Box Info section of the MIO Console does not register the presence of the box.

   • This symptom is also present if a copy of the "MobileIO Driver" file is installed in the wrong location (see below).

   • To correct this condition, make sure the "MobileIO™ Enabler" file is installed in the "System Folder:Extensions" folder and then reconnect the Mobile I/O to the computer.

2. If the "MobileIO Driver" file is not installed in the "System Folder:Extensions" folder, the MIO Console will not start-up and the ASIO driver will not connect properly.

   • To correct this condition, make sure the "MobileIO Driver" file is installed in the "System Folder:Extensions" folder and then reconnect the unit to the computer.

3. If you have accidentally put a copy of the "MobileIO Driver" file in the same folder with the MIO Console application or with any of the ASIO driver files, the symptoms will be similar to what you see with condition #1. The Console or the ASIO driver will appear to run properly, but it will not see the box connected to the computer.

   • To correct this condition, check for a copy of the "MobileIO Driver" file in any location other than the "System Folder:Extensions" folder. If a copy of the file exists, delete it from your disk or, at the very least, ensure that it is not in the same folder with the MIO Console application or any of the ASIO Driver files.

4. If the "Mobile IO™ ASIO" file is not installed in the appropriate folder for each of your hosts, the corresponding host will not show the Mobile I/O as an available ASIO driver. Most ASIO hosts use a folder called "ASIO Drivers" located in the same folder as the host application. If your specific host does not use this convention, you will have to check the documentation that accompanied your host or contact the manufacturer of the host for assistance.

   • Note that you need to install a copy of the "Mobile IO™ ASIO" file in the appropriate folder for *each* host you will use on your computer. In general, ASIO Drivers are not shared between hosts.

If the software is installed properly, check the next possibility.

THE FIREWIRE BUS DID NOT RESET CORRECTLY

When a device is plugged into the FireWire bus, a FireWire bus reset occurs automatically. The bus reset interrupts bus activity and reconfigures the bus so that all devices on the bus become aware of all the other devices on the bus. Sometimes the reset does not complete successfully, and the bus becomes partially hung. In this case, the "FireWire" indicator on the front panel of the Mobile I/O will not be illuminated. When the "FireWire" indicator on the front panel is not illuminated, the Mobile I/O cannot transport audio over the FireWire bus.

Generally, this condition can be fixed by disconnecting the Mobile I/O from the bus and reconnecting it.

If the disconnect/reconnect cycle does not fix the problem, another device on the bus may be interfering with the proper operation of the bus. If you have other devices on the bus, try disconnecting them from the bus and only using the Mobile I/O.

If removing other devices from the bus solves the problem, it is likely that there is a problem with either one of the devices you removed or with one of the cables connecting the devices. You'll need to isolate the problem component.

If removing the other devices from the bus does not fix the problem, check the next possibility.

The FireWire Cable is Bad

Metric Halo provides two high-quality overspec'ed FireWire cables for use with Mobile I/O and we recommend you use them. For various reasons you may decide to use other cables than the ones provided by Metric Halo. Under ideal circumstances all FireWire cables will provide years of service. However, cables will and do go bad. Cable failures can be difficult to track down. If you are experiencing problems with connecting or bus powering Mobile I/O you should try swapping the cable with another known-good cable.

If the FireWire cable is not the source of the problem, check the next possibility.

The FireWire Hardware has been Damaged

If all else fails, it may be that the FireWire hardware on either the Mobile I/O or the computer has been damaged. While this is an exceptionally rare occurrence, it is a possibility. The FireWire hardware can be damaged in the following ways:

1. If you insert a FireWire cable into a port upside down, it will damage the FireWire port and/or the connector. It is difficult to insert the connector upside down, but it is possible to force it. Never force a FireWire connector!

2. It requires significant pressure, but it is possible to force a FireWire connector over a male XLR connector pin. If you do this, the connector will be shorted and it will destroy the port on the other end. Again, never force a FireWire connector.

3. Some devices that are bus-powerable and conform to the IEEE1394 standard will return power to the remote FireWire port if a power ground fault occurs. If the remote port is protected against this situation, nothing will happen. If the device does not use bus power, nothing will happen. But, if the device is fully compliant, uses bus power, and the remote device is not protected and supplies a high enough voltage on the bus, the remote device port will self-destruct.

If the FireWire hardware on the computer has been damaged, it will not communicate with any FireWire devices. Be sure that you are not checking this case with a bad cable, as a bad cable can make it seem like the FireWire

hardware has failed since it will consistently keep devices from connecting properly to the computer. If the computer is damaged, you will need to contact the manufacturer for a repair or, as a stopgap measure you can use a third-party FireWire adapter card.

If the FireWire hardware on the Mobile I/O has been damaged (an event that has never occurred), the Mobile I/O will not communicate with any other devices. In this case, please contact Metric Halo support for help in getting your Mobile I/O hardware repaired.

ADAT DOES NOT APPEAR TO LOCK

The Mobile I/O ADAT input clock is automatically phase synchronized to the incoming ADAT stream. If you use the ADAT as the clock source for Mobile I/O, you will always be able to lock to the input stream as long as the ADAT output of the external device feeding the Mobile I/O is correctly formatted and does not have excessive jitter.

An example of a device that has excessive jitter due to an ADAT implementation problem is the Digidesign ADAT bridge. Digidesign's more recent implementations (e.g. the ADAT interface in the Pro Tools HD system) do not exhibit this problem.

If you use a clock source other than the ADAT input, the input stream lock will depend on the detailed properties of the external ADAT source's ADAT-to-clock synchronization. If the synchronization does not meet Mobile I/O's requirements you will not get a clean ADAT signal, even if the two devices are nominally synchronized. The ADAT implementation from Alesis works fine in all cases, but other devices may not. In the case that you want to use ADAT input and synchronization is not working, select ADAT as your clock source and your sync problems will disappear. This discussion does not generally apply to the ADAT output.

MIO CONSOLE IS NOT SYNCHRONIZED WITH THE HARDWARE

If you quit MIO Console or reboot the Mobile I/O hardware while MIO Console is running, the settings of the Console will become desynchronized with the hardware.

If the current configuration of the Console is what you want, the easiest way to reconfigure the hardware is to save the state of the console to a console snapshot file and then reload the state from the file.

If you want to reset the console state to the state of the hardware, there is no good method available at this time, if you did not save the state of the Console to disk before quitting the application. We recommend that you save the current state of the console before quitting the MIO Console application, so that you can recall the configuration at a later time.

Metric Halo is currently investigating more complete solutions to this issue.

WHEN I SWITCH TO MIO CONSOLE, PLAYBACK STOPS

Some ASIO Hosts will automatically stop playback when the host is switched to the background. This occurs when you switch to MIO Console (or any other application). All of these hosts have an option to continue playing in the background; you need to ensure that the option is enabled. In addition to the host's own preference, many of the hosts that support MIDI sequencing will stop playing if the MIDI system (OMS or FreeMIDI) has been configured to release the MIDI port when in the background. While it is a bit confusing, you need to ensure that the MIDI system is configured to not share or release the MIDI port.

A more subtle issue is that MIO Console has an option to optimize its drawing operations which can be interrupted by background application activity. If the "**Allow Background Apps to Run**" item in the MIO Console's **Edit** menu *does not* have a diamond mark (◊) next to it, the console will lock other applications out while it is the active application. This is not a problem for most ASIO Hosts, but it does affect Digital Performer, which will stop playing after 10-20 seconds. To resolve this issue, make sure that the "**Allow Background Apps to Run**" item in the MIO Console's **Edit** menu *does* have a diamond mark (◊) next to it.

WHERE ARE THE {EXTENSIONS, ASIO DRIVERS} FOLDERS

The **Extensions** folder is a system wide location for all third-party extensions to the Mac OS. It is located within the currently active **System Folder**. You can generally find it by using Sherlock to find files named "Extensions". Con-

sult the documentation for your Macintosh for more information. Please note that localized versions of the Mac OS use different, translated names for both the **Extensions** folder and the **System Folder**.

The **ASIO Drivers** folder, by convention, is the location that each ASIO host uses to manage ASIO drivers. Most hosts follow the convention that ASIO drivers are stored in a folder named "ASIO Drivers" that is located in the same folder as the host application. Each host has its own **ASIO Drivers** folder; they are not shared. You need to install the Mobile IO ASIO driver in the **ASIO Drivers** folder for each host you want to use with Mobile I/O. If your host does not follow the convention, you will have to consult the documentation provided with the host to determine where to install the ASIO driver.

GROUND LOOPS

Audio systems, in general, are susceptible to ground loop problems. Digital Audio Interfaces for computers are even more susceptible to grounding issues since they must interface with the computer's system ground, which tends to be much more dirty than the ground used by audio gear. By taking care when you connect the various components of your audio system you can avoid the hums, buzzes, and noises that characterize ground loops and other grounding problems.

First of all, most grounding issues go away if you utilize balanced interconnects between your audio gear. Balanced interconnects inherently reject ground differentials and common mode interference introduced by grounding problems. Balanced connections are not much more expensive than unbalanced connections and solve so many problems that if both ends of the connection support balanced interconnect, you should not even consider using unbalanced cables.

You may get the idea that we hate unbalanced connections. You're right. We do. You should too.

If you have to use unbalanced connections, or if any ground-related problems remain, you will find that the key to the issue is ensuring that you have a common hard ground between all the gear that you are interfacing. This is commonly referred to as a technical ground. A technical ground is charac-

terized by a consistent low impedance path between each device and a common reference ground, ideally connected directly to earth ground. The above is sometimes difficult due to electrical wiring problems in the house, studio, or stage you are using. In the extreme case, you may need to hire a qualified electrician to untangle and correct electrical service problems in your working environment.

Unbalanced connections are a fact of life when interfacing with guitar amps, and, paradoxically, guitar amps are extremely sensitive to grounding issues since they use so much gain to achieve the effect of a "Guitar Amp". If you will be interfacing with guitars and guitar amps, you need to be very careful about grounding.

Common electrical wiring approaches to residential installations, and sub-par studio and stage installations use daisy-chained grounds for ease of installation and economy. Unfortunately, daisy-chained grounds can introduce signficant ground differentials between sockets, and these differentials can vary depending on other loads (like refrigerators, TV's and other household appliances) on the circuit.

Other problems with electrical service installations are improper wiring of power phases to the three-phase service and improper connections between the safety ground and hot legs of the three-phase service. These types of problems tend to be characterized by loud 60Hz hums in the audio system. Unfortunately, these types of problems extend well beyond noise in your audio system to genuine safety hazards. If you determine that your electrical wiring has problems beyond a simple daisy-chained ground, you should consult a licensed electrician immediately, as ignoring these problems can damage either you or your gear.

If you do not have a well implemented technical ground, you will want ensure that all of the devices in your audio system are plugged into the same phase and same ground. You can generally accomplish this by running all your gear off of the same socket (using a power strip or power conditioner) if your gear uses less power than is supplied by a single circuit from your premises wiring (generally 10-15 amps in residential installations and 20 amps in commercial installations).

It is usually a bad idea to put some devices in your system on a power conditioner and other devices on a separate strip, socket or conditioner, unless you have a technical ground. The power conditioner can introduce a ground differential.

The power supply provided with Mac laptops does not have a hard ground. This means that if the laptop is plugged in, it will dump high frequency buzz into the ground. That ground is shared with the Mobile I/O Firewire cable. If Mobile I/O will be connected unbalanced to other audio gear, the ground buzz can contaminate the signal if the Mobile I/O is not hard-grounded to the same ground as your other audio gear. To hard ground the Mobile I/O you will need to use a 3-pin power cable on the Mobile I/O power supply and power the Mobile I/O with the power supply. Plug the 3-pin IEC power cable into the same circuit and same ground as your other gear.

On the other hand, if you are encountering ground loop problems while operating with the Mobile I/O's power supply, you may find that lifting the Mobile I/O's ground resolves the problem. This can be accomplished by using a 2-Pin IEC cable (without the third ground pin), or by using a ground lift block (generally available in hardware stores, also known as a 3 pin to 2 pin converter). In general, it is better to resolve the fundamental grounding problems in your system, but this is a quick fix that may help. There are no hard and fast rules for solving this type of problem other than fixing the fundamental grounding issues, so if you go this route, you will have to experiment with lifting various grounds in your system until you find the magic combination. Or switch to balanced interconnects.

Finally, the Apple Cinema Display has a known issue with its backlight dimmer. If you run the Apple Cinema Display with its backlight at anything other than full brightness, the backlight dimmer will introduce a midrange buzz into the system ground which will appear in unbalanced interconnects (input and output) with Mobile I/O. This issue affects other devices that connect to the computer's system ground. The work around is to run the display at full brightness, or use balanced interconnects.

DIGITAL GRUNGE

It is possible for the ASIO driver to fall out of sync with the hardware. This is an unlikely occurrence and Metric Halo is actively investigating the root cause. It appears to affect some hosts more than others and is more likely to happen if the system is heavily loaded. If it does happen, the audio being transpoerted will have an aliased, bit-crusher quality of digital distortion to it. You will need to reset the driver to fix this problem. Each host has a different approach to resetting the driver – check your host's documentation for details.

FIRMWARE UPDATE PROBLEMS

For details on updating the firmware of Mobile I/O refer to Appendix 1.

It is possible for firmware updates to "not take". This appears to be related to DSP loading issues in the Mobile I/O, other devices on the FireWire bus, and the state of the FireWire system software on the Mac. If you have problems with updating the firmware, try the following procedure:

1. Remove all devices from the FireWire bus
2. If your Mobile I/O is using external power, disconnect the power
3. Reboot your computer
4. Attach the external power supply to the Mobile I/O **while holding down the front panel Mute button;** this will boot the Mobile I/O into the safety boot firmware
5. Connect the Mobile I/O to your computer
6. Run the firmware updater (MIOBootInstaller at this writing)

Since Mobile I/O implements safe-boot and safe firmware update, you should always be able to use this procedure to update the firmware, even if something goes horribly wrong (like losing power during an update).

BUS POWERING MOBILE I/O

If you are bus powering the Mobile I/O, there is a possibility that you have overloaded the power rating of the power source.

NOT ENOUGH POWER ON THE BUS

While all Macintosh computers with built-in FireWire supply bus power, some models do not provide enough power on the bus to power Mobile I/O.

If this is the case, you will generally find that the Mobile I/O will boot on initial connection, but will then lose power or will reboot repeatedly after a short period of operation.

Some Mac models provide enough power if they are plugged into the wall, but will not provide enough power while running on batteries. If the computer does not provide enough power, you will need to use an external power source with Mobile I/O.

The external power supply provided with Mobile I/O is the perfect solution if you are using Mobile I/O in an environment where AC power is available. The external power supply will actually provide power to the bus and can be user to power other bus-powered peripherals (see "**Other Bus Powered Devices**" on page 92).

If AC power is not available, you will need to use an external battery-based power source to power Mobile I/O. Any source that provides 9V-30V and can support 12-15W of power consumption will work well with Mobile I/O. Check with Metric Halo for specific recommendations.

When using an external battery source, DO power Mobile I/O directly from the battery – not through an inverter. DON'T power the computer with an external battery and use the computer to power Mobile I/O; doing so will not resolve your bus power problems, and it will give you more limited run times. If you need to use the external battery with the computer use two batteries or split the DC supply at the battery and power both the Mobile I/O and the computer.

HIGH POWER MODE
When the Mobile I/O is in High Power mode, the analog power rails are run at a higher voltage than in standard mode. High Power mode provides greater output headroom and reduced distortion at high (pro-audio) analog output levels, but requires substantially higher (>5W) supply power. Most currently available computers are not able to provide enough current over the bus to power Mobile I/O in High Power mode. The current transient generated when the unit is switched to High Power mode can cause damage to machines that have low bus power limits. We **STRONGLY** recommend powering the Mobile I/O from an external power supply if you want to use High

Power mode. Metric Halo accepts no responsibility for any damage that may occur if you enable High Power mode while Mobile I/O is running on bus power.

SCREEN BACKLIGHT ISSUES

When Mobile I/O is attached to FireWire bus without any external power applied, there is a small current surge as the Mobile I/O power supply charges up. Depending on the firmware revision and model type of your PowerBook and what other devices are attached to or installed in your computer, this inital connection may trick the power manager in the computer into thinking that it is about to lose power. In this circumstance, the power manager will shut-down the computer's backlight. The computer has not crashed or gone to sleep; only the backlight has been shut down.

There does not seem to be anyway to turn the backlight back on without putting the machine to sleep. The workaround for this condition is to put the machine to sleep (either by closing the lid or pressing <command><shift>0 [zero]) and then waking it up again with the Mobile I/O still attached. This will not happen if the Mobile I/O is externally powered.

With early Titanium PowerBooks, especially with early versions of Apple's firmware, spinning up the CDROM or hard disk while bus powering Mobile I/O may trigger the screen backlight issue. We recommend that you remove all media from the CD/DVD drive and set the hard disk to never sleep when you are bus powering Mobile I/O with these machines.

OTHER BUS POWERED DEVICES

Mobile I/O consumes enough power that it is very unlikely that you will be able to successfully bus power Mobile I/O and any other bus-powered device (except for a hub) from the same computer. If you plan on using other bus-powerable devices with your computer, you will need to either self-power your other devices or self-power the Mobile I/O. It is probably best to use the Mobile I/O's power supply in this situation since Mobile I/O will then provide approximately 30 Watts of power to the bus (roughly 3x what most Portable Mac's will supply). This will allow you to power all the rest of your devices without any concern of running out of power.

Appendix I – Updating your Firmware

The Mobile I/O is a complex device with a complex DSP–based signal processing and control architecture. One of the major strengths of Mobile I/O's design is that the operating system of the box can be upgraded at any time by updating the firmware. The firmware provides data to the hardware upon system boot that configures both fundamental aspects of the hardware and the operating system for the box. This data is stored in a memory device on the Mobile I/O motherboard. The data can be updated at any time, but it will be maintained indefinitely, even without any power being applied to the Mobile I/O.

Since the hardware itself can be reconfigured by the firmware, this approach allows Metric Halo to make major enhancements to Mobile I/O without any physical changes to the hardware. In the past we have used software deployed firmware updates to increase the FireWire access speed, provide independent heaphone channels, and improve the converter sound quality over its already exceptional character.

Since the firmware updates exist simply as data, they can be sent to you in a variety of ways, whether via CD, email or download from our website. Metric Halo provides a firmware update tool that allows you to download an updated firmware image to the Mobile I/O from your computer. The following section describes how to use the current version of the firmware update tool, called **MIOBootInstaller.**

You may have had the experience of updating the firmware for your computer in the past. As you may know, this can be a stressful procedure, since there is a moment while the old firmware is being replaced by the new firmware, and if the process is interrupted you may be left with no firmware at all. Metric Halo has addressed this issue with a "safe firmware update" procedure. The Mobile I/O uses a dual-boot procedure. The first boot happens in the first 100ms (about 0.1 seconds) and has been extensively tested. It is smart enough to do two things:

1. It can boot the secondary boot image
2. It can update the secondary boot image over the FireWire bus

Actually, the primary boot firmware is much smarter than that. The box is completely functional on the primary boot, but all of the more advanced features of the box are enabled by the second boot. The firmware revision of the primary boot is **1.1.0**.

As soon as the primary boot image has booted, it checks the secondary boot image, and if the secondary boot image is installed and not corrupted in any way, the system immediately boots the secondary image. If the secondary image is corrupted or if you have held down the front-panel **Mute** button during the initial boot process, the Mobile I/O will not boot the secondary image and will stay in "**Safety Boot Mode**". This is a mechanism you can use if you install firmware that has problems and you need to back up or install a newer image.

The safe boot mechanism allows you to back out of a firmware update if you find that the new firmware has some problem that was not present in a previous version of the firmware. Metric Halo support may ask you to do this during troubleshooting if you encounter any problems.

In the future, Metric Halo may change the methods or tools used to find, download, and accomplish firmware updates. If the tools change, they will be accompanied by an updated version of this Appendix.

Installing a firmware update

In order to install a firmware update, follow these steps:
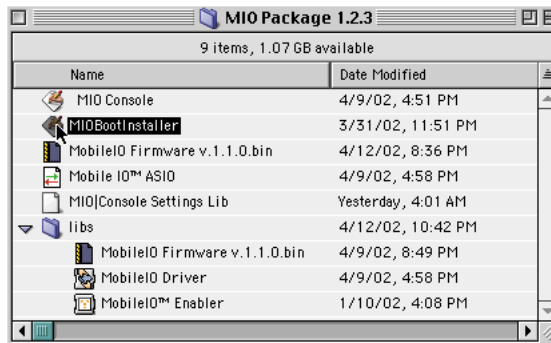
Figure 65: MIO Package

1. If there is an associated driver update, place the MobileIO Driver and MobileIO Enabler in your extensions folder inside your system folder.
2. Reboot your computer.
3. Make sure your Mobile I/O is powered up and connected to your computer.
4. Double click on MIOBootInstaller.

You will see the following:

Figure 66: MIOBootInstaller Window
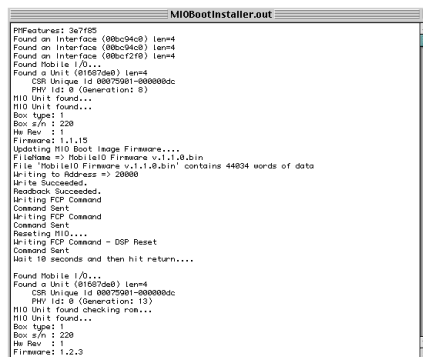
MIOBootInstaller will find your Mobile I/O on the FireWire bus and begin sending commands to it. When MIOBootInstaller is finished updating the

95

firmware, it will say "Wait 10 seconds then hit return..." After you hit return, MIOBootInstaller will find your Mobile I/O again and display your unit's serial number and the firmware revision which should match the firmware revision identified with the package.

Copy the Mobile I/O ASIO Driver into the ASIO folders of the audio applications you want to use, reboot your computer, and you're ready to go.

Rolling back your firmware

If you find that you have problems with any given release, you can always go back to a previous release by downloading a package from

http://www.mhlabs.com.com/mio

and following the update instructions in that package.

Index