

MOTU 896HD™

User's Guide for Windows



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SAFETY PRECAUTIONS AND ELECTRICAL REQUIREMENTS

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

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

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

WARNING: IF NOT PROPERLY GROUNDED THE MOTU 896HD COULD CAUSE AN ELECTRICAL SHOCK.

The MOTU 896HD is equipped with a three-conductor cord and grounding type plug which has a grounding prong, approved by Underwriters' Laboratories and the Canadian Standards Association. This plug requires a mating three-conductor grounded type outlet as shown in Figure A below.

If the outlet you are planning to use for the MOTU 896HD is of the two prong type, DO NOT REMOVE OR ALTER THE GROUNDING PRONG IN ANY MANNER. Use an adapter as shown below and always connect the grounding lug to a known ground. It is recommended that you have a qualified electrician replace the TWO prong outlet with a properly grounded THREE prong outlet. An adapter as illustrated below in Figure B is available for connecting plugs to two-prong receptacles.

Figure A

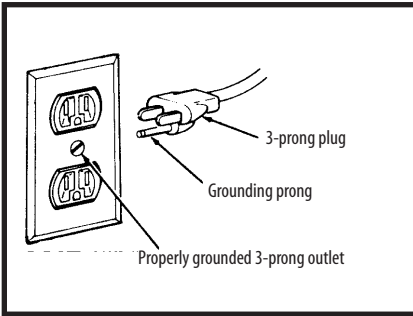
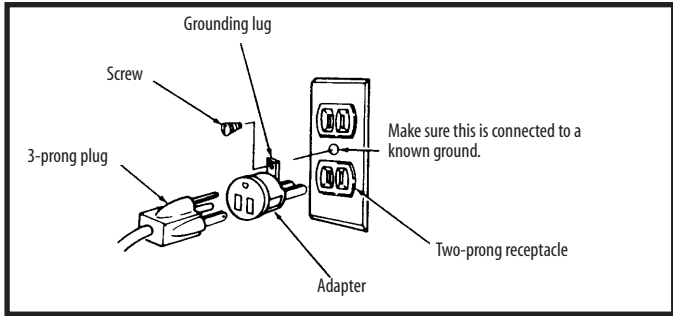


Figure B



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

If you are not sure that your outlet box is properly grounded, have it checked by a qualified electrician. NOTE: The adapter illustrated is for use only if you already have a properly grounded two-prong receptacle. Adapter is not allowed in Canada by the Canadian Electrical Code. Use only three wire extension cords which have three-prong grounding type plugs and three-prong receptacles which will accept the MOTU 896HD plug.

IMPORTANT SAFEGUARDS

1. Read instructions - All the safety and operating instructions should be read before operating the MOTU 896HD.
2. Retain instructions - The safety instructions and owner's manual should be retained for future reference.
3. Heed Warnings - All warnings on the MOTU 896HD and in the owner's manual should be adhered to.
4. Follow Instructions - All operating and use instructions should be followed.
5. Cleaning - Unplug the MOTU 896HD from the computer before cleaning and use a damp cloth. Do not use liquid or aerosol cleaners.
6. Overloading - Do not overload wall outlets and extension cords as this can result in a risk of fire or electrical shock.
7. Power Sources - This MOTU 896HD should be operated only from the type of power source indicated on the marking label. If you are not sure of the type of power supply to your location, consult your local power company.
8. Power-Cord Protection - Power-supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them. Pay particular attention to cords and plugs, convenience receptacles, and the point where they exit from the MOTU 896HD.
9. Lightning - For added protection for the MOTU 896HD during a lightning storm, unplug it from the wall outlet. This will prevent damage to the MOTU 896HD due to lightning and power line surges.
10. Servicing - Do not attempt to service this MOTU 896HD yourself as opening or removing covers will expose you to dangerous voltage and other hazards. Refer all servicing to qualified service personnel.
11. Damage Requiring Service - Unplug the MOTU 896HD from the computer and refer servicing to qualified service personnel under the following conditions.
 - a. When the power supply cord or plug is damaged.
 - b. If liquid has been spilled or objects have fallen into the MOTU 896HD.
 - c. If the MOTU 896HD has been exposed to rain or water.
 - d. If the MOTU 896HD does not operate normally by following the operating instructions in the owner's manual.
 - e. If the MOTU 896HD has been dropped or the cabinet has been damaged.
 - f. When the MOTU 896HD exhibits a distinct change in performance, this indicates a need for service.
12. Replacement Parts - When replacement parts are required, be sure the service technician has used replacement parts specified by the manufacturer or have the same characteristics as the original part. Unauthorized substitutions may result in fire, electric shock or other hazards.
13. Safety Check - Upon completion of any service or repairs to this MOTU 896HD, ask the service technician to perform safety checks to determine that the product is in safe operating conditions.

ENVIRONMENT

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

AVOID THE HAZARDS OF ELECTRICAL SHOCK AND FIRE

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

INPUT

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

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

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

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

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

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

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



Quick Reference: 896HD Front Panel

The light that is illuminated here tells you which bank (analog in, ADAT in, or ADAT out) you are monitoring with the programmable meter bank to the right.

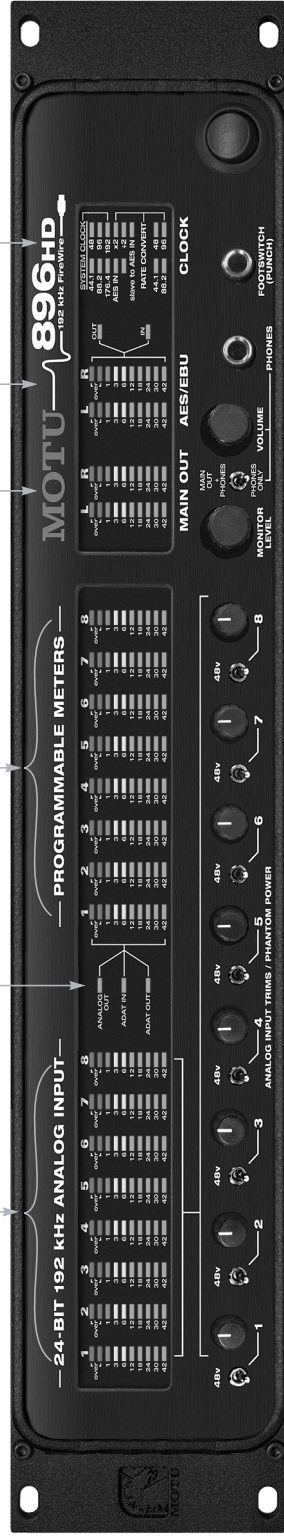
These 10-segment level meters are dedicated to the 896HD's eight analog inputs. The top red 'over' LED lights up when the signal reaches full scale — for even just one sample — and remains illuminated until you clear it in the software. The second 'over' LED below only lights up momentarily so that you can continue to adjust level even after clipping has just occurred. These level meters can also show you the current volume setting for the 896HD's four on-board mix buses. To view the bus volume settings, push the MONITOR LEVEL knob.

This bank of level meters can be programmed to display one of three different banks: analog output, ADAT optical input, or ADAT optical output. Use the MOTU FireWire Audio Console to choose which bank you'd like to view with these meters. The LEDs to the left show which bank you are currently monitoring. These meters can also provide stereo output metering for the 896HD's on-board mix buses (by pressing the MONITOR LEVEL knob).

These meters can be programmed to display either AES/EBU input or output. Use the MOTU FireWire Audio Console to choose. The LEDs to the right indicate what you are currently monitoring.

These lights indicate the global sample rate at which the MOTU 896HD is operating. Use the MOTU FireWire Audio Console to set the sample rate or to choose an external clock source, from which the sample rate will be set. When no sample clock is currently present, these lights flash. For example, if you've set the MOTU 896HD to slave to an external clock, such as ADAT, but there is no clock signal currently being detected, these lights will flash.

These meters display the level for the 896HD's main outs.



This toggle switch provides phantom power for a condenser mic plugged into this input. Up is on.

These eight trim knobs provide gain adjustment within a range of about 42dB for each analog input (when the 3-way switch on the rear panel is set to either MIC or LINE. If the 3-way switch is set to +4/FIXED, the corresponding front-panel trim pots is bypassed). Because each input also has a microphone preamp, you can plug just about anything into them: a microphone, a guitar, a synth — you name it.

For a microphone or unamplified instrument pickup, set the rear-panel 3-way switch to MIC, plug in your mic (XLR or quarter-inch plug) flip on 48V phantom power (if necessary) and use the trim knob as needed to adjust the level.

For +10dB (unbalanced) inputs (like synths) or +4 (balanced) signals, set the rear panel 3-way switch to LINE and use the trim knob to adjust the level. You can use either an XLR or quarter-inch plug. (Note: the complete trim range, from all the way down with the MIC setting to all the way up with the LINE setting is around 55dB total. The MIC setting provides a 40dB range (approximately -37dBu to +5dBu) and the LINE setting offers a 30dB range (approximately -16dBu to +15dBu) with some overlap between them.)

If you have +4 inputs for which you'd like to maintain unity gain (for levels up to +18dBu), set the rear-panel 3-way switch to +4/FIXED. You can use either an XLR or quarter-inch plug.

The MOTU 896HD provides four independent mix buses. Via the included CueMix Console software, these four buses can be programmed to mix any combination of the 896HD's 18 inputs to any output pair of your choosing. The MONITOR LEVEL knob lets you control the volume of each mix from the front panel. Push the knob repeatedly to cycle among the four different mixes (1, 2, 3 or 4). When you do so, the analog input meters (1-4) show the current volume setting with the red clip light illuminated over the bus you are currently adjusting when you turn the knob. The programmable meter bank provides live metering for each bus.

If the toggle switch is down, the volume knob affects the headphone out (and the main outs operate at a fixed volume). If the toggle switch is up, the volume knob controls the main outs and the headphone out. Analog input meters 7-9 display the current headphone (7) and main out (8) volume setting as you turn the volume knob. Push the volume knob to cycle through the various settings for the programmable meters (both the 8-channel bank and the stereo AES/EBU meters).

This is a standard quarter-inch stereo headphone jack. From the factory, it matches the signal going to the main outs. However, via software, you can program it to serve as its own independent output.

Connect a standard foot switch here for hands-free punch-in and punch-out during recording. For details about how to set this up, see "Enable Pedal" on page 39.

FireWire is a "plug-and-play" protocol, that means that you can turn off the MOTU 896HD and turn it back on (or even unplug it) without restarting your computer. Keep in mind, however, that if you wish to change the MOTU 896HD's settings in the MOTU FireWire Audio Console, the MOTU 896HD needs to be plugged in and switched on.

Quick Reference: 896HD Rear Panel

These AES/EBU connectors can handle any supported sample rate up to 96 kHz, and they are also equipped with a sample rate converter so you can input or output at a different rate than the 896HD. For details, see "Syncing AES/EBU devices" on page 28. At the 4x sample rates, (176.4 and 192kHz), AES/EBU is disabled.

These two XLR jacks serve as the MOTU 896HD's main analog outputs. You can connect them to a set of powered studio monitors and then control the volume from the front panel Volume knob. At the 4x sample rates (176.4 or 192kHz), these jacks always mirror the headphone output.

If you are using the MOTU 896HD with an ADAT, use this standard ADAT SYNC INPUT to connect the MOTU 896HD to the end of your ADAT sync chain. For example, if you have three ADATs, chain the ADATs in the usual fashion (SYNC OUT to SYNC IN, etc.), and then connect the last ADAT'S SYNC OUT to this SYNC IN. This connection allows you to make sample-accurate audio transfers between AudioDesk (or other sample-accurate software) and the ADATs. If you have a MOTU MIDI Timepiece AV or Digital Timepiece, make it the master of the ADAT SYNC chain so that you can control everything from AudioDesk (or your other MIDI Machine Control compatible software).

The MOTU 896HD's eight analog outputs are XLR connectors with +4/-10 switchable output (600 Ohm impedance). They are equipped with 24-bit, 128x oversampling enhanced multibit A/D converters capable of 192kHz recording.



The 896HD power supply is switchable between 110V and 220V operation. It should already be set to the proper voltage for your country, but you can check the setting and adjust it if necessary with the red switch just to the left of this power cord receptacle (on the side of the metal chassis).

Use the word clock input and output for digital transfers with devices that cannot slave to the clock supplied by their digital I/O connection with the 896HD.

These ADAT optical I/O connectors can be connected either to an ADAT or any ADAT-compatible "lightpipe" device (such as a digital mixer). ADAT optical supplies eight channels of 24-bit digital I/O at 44.1 or 48kHz. It carries 4 channels of 24-bit digital I/O at 88.2 or 96kHz (it is disabled at the 4x sample rates.)

These eight analog inputs are Neutrik™ combo connectors that accept either an XLR plug or a quarter-inch plug. They have 24-bit, 64x oversampling converters. Each input is equipped with a 3-way input level switch with three settings:

LINE: Use this setting for +4dB or -10dB inputs, such as synthesizers or consumer audio equipment. Adjust the input level as needed with the corresponding front panel Trim control and level meter for this input. This setting offers a trim range of around 30dB (approximately -16dBu to +15dBu).

+4 / FIXED: Use this setting for +4 balanced inputs for which you do not wish to modify the gain. This position disengages the front panel trim knob. This setting also provides slightly more attenuation than the LINE setting, allowing levels up to +18dBu.

MIC: This setting feeds the input signal (XLR or quarter-inch) to the built-in preamp for additional gain. Use it for any microphone or unamplified instrument pickup. Engage phantom power for condenser mics with the front-panel 48V switch (up is on). Use the corresponding front-panel trim knob and level meter to adjust the input level as needed. This setting offers a trim range of around 40dB (approximately -37dBu to +54dBu).

Connect the MOTU 896HD to the computer here using the standard 1394 FireWire cable provided with your MOTU 896HD. Use the extra FireWire port to daisy-chain up to four 896HD's to a single FireWire bus. You can also connect a MOTU 828, FireWire hard drive or other FireWire device. For details, see "Connecting multiple MOTU FireWire interfaces" on page 33.

Quick Reference: MOTU FireWire Audio Console

Determines the clock source for your 896HD. If you're just using the analog ins and outs, set this to 'Internal'. The other settings are for digital transfers via AES/EBU or external synchronization via the ADAT SYNC in port.

If you have a foot switch connected to the 896HD, these settings let you map the foot switch to any computer keyboard key for both the up and down position. For details about how to set this up, see "Enable Pedal" on page 39.

Check this option if the audio software you are using with the 896HD does not support Windows WDM drivers and instead only supports legacy MME (Wave) drivers. When checked, this option makes all of the 896HD inputs and outputs available to legacy Wave-driver compatible audio software.

- Phones
- Analog 1-2
- Analog 3-4
- Analog 5-6
- Analog 7-8
- Main Outs
- AES/EBU
- ADAT 1-2
- ADAT 3-4
- ADAT 5-6
- ADAT 7-8

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

Lets you choose what to monitor with the 896HD's programmable front panel meters.

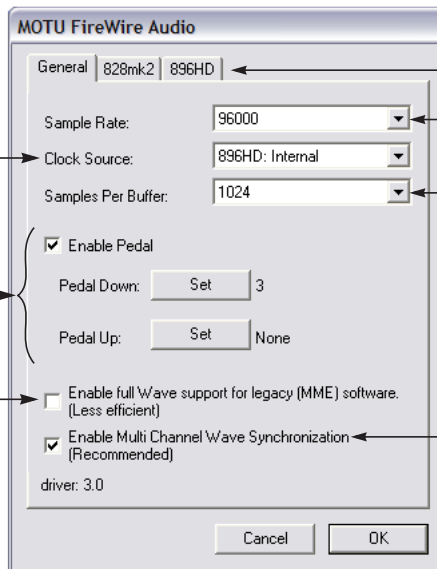
If you are running the 896HD interface at a 2x or 4x sample rate (88.2 to 192kHz), this option appears at the bottom of the interface settings. It lets you choose a word clock output rate that either matches the global sample rate (e.g. 192kHz) or forces a word clock output rate of either 44.1 or 48kHz.

Takes the interface off line. Inputs and outputs are not available to audio applications, and bandwidth on the FireWire bus is relinquished, freeing it up for other devices.

How to access these settings

There are several ways to access these settings:

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

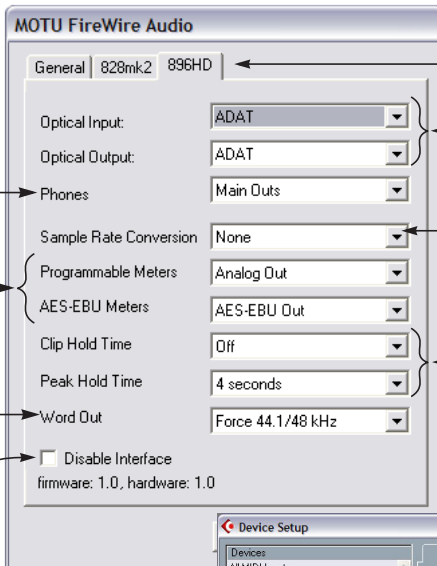


Click the tabs to access general MOTU FireWire interface settings or settings specific to the 896HD (or other connected interface.)

Choose the global sample rate for the system here.

Choosing a smaller setting here reduces the delay you may hear when listening to live input that you are running through effects plug-ins in your software. But lower settings also increase the strain on your computer. For details, see "Samples Per Buffer" on page 38.

This option should always be left on (checked). There are only a few rare cases in which you would want to turn it off. For details, refer to the MOTU tech support database at www.motu.com.

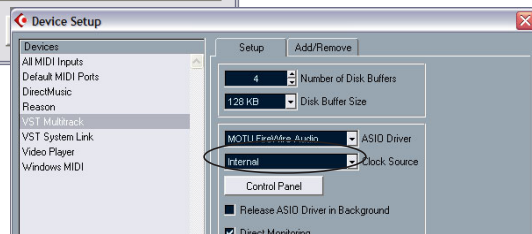


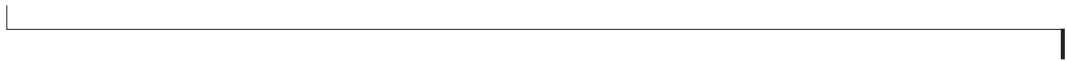
Click the 896HD tab to access these settings.

Enables or disables the 896HD's ADAT optical I/O. Turning them off frees up FireWire bandwidth.

Provides several options for the MOTU 896HD's AES/EBU sample rate conversion. See, "Syncing AES/EBU devices" on page 28.

The *Clip Hold Time* controls how long the top-most red LED remains illuminated after clipping. Choose 'Infinite' if you want to be able to clear the LED from Digital Performer. The *Peak Hold Time* controls how long the highest illuminated LED remains lit before going dark.





CHAPTER 1 About the 896HD

OVERVIEW

The 896HD is a computer-based hard disk recording system for Mac OS and Windows that provides 18 separate inputs and 22 separate outputs (at 44.1 or 48kHz), including separate main outs and headphone out. All inputs and outputs can be accessed simultaneously. The 896HD consists of a standard 19-inch, double-space, rack-mountable I/O unit that connects directly to a computer via a standard IEEE 1394 FireWire™ cable. The 896HD offers the following:

- Operation at 44.1, 48, 88.2, 96, 176.4 or 192 kHz
- Eight 24-bit analog outputs individually switchable between +4 and -10dB operation
- Eight 24-bit analog inputs equipped with Neutrik “combo” jacks and independent 3-way level switch for MIC, LINE or +4/FIXED inputs
- Eight-channel ADAT optical digital I/O
- AES/EBU with sample rate conversion
- Two extra analog main outs
- Eight mic preamps (one on each input)
- Independent 48V phantom power for each input
- Independent front-panel trim for each input
- Sample-accurate ADAT sync input
- Word clock input and output
- Two FireWire jacks for chaining multiple units
- Foot switch input for hands-free punch-in/out
- Front-panel Headphone jack
- Main volume knob (for headphone + main outs)
- CueMix™ DSP no-latency monitoring

- 10-segment LED level meters for each input
- 10-segment programmable LEDs for analog output, ADAT input or ADAT output
- 10-segment programmable LEDs for AES/EBU input or output
- Switchable power supply (110V or 220V)

With a variety of I/O formats, mic preamps, no-latency monitoring of live input and synchronization capabilities, the 896HD is a complete, portable “studio in a box” when used with a Macintosh or Windows computer. A WDM driver is included for audio applications running under Windows Me/2000/XP. Also included is an ASIO driver for multi-channel operation with any Windows audio software that supports ASIO drivers. The 896HD’s GSIF driver provides low-latency multi-channel I/O with Tascam GigaSampler™ (and the other Giga family products).

THE 896HD I/O REAR PANEL

The 896HD rear panel has the following connectors:

- Eight 24-bit XLR analog outputs, each equipped with an independent 2-way output level switch (+4 or -10dB)
- Eight 24-bit Neutrik “combo” (XLR + balanced quarter-inch) analog inputs, each equipped with an independent 3-way level switch (MIC, LINE, +4/FIXED), mic preamp, front-panel 48V phantom power switch, and front-panel trim knob
- One set of ADAT optical ‘light pipe’ connectors (8 channels of ADAT optical input and output at 44.1/48kHz and 4 channels at 88.2/96kHz)
- AES/EBU input and output

- Two XLR main analog outputs with volume knob (on the front panel)
- One 9-pin ADAT SYNC IN connector
- BNC word clock input and output
- Two 1394 FireWire jacks

18 inputs and 22 outputs

All 896HD inputs and outputs can be used simultaneously, for a total of 18 inputs and 22 outputs at 44.1/48kHz:

Connection	Input	Output
24-bit 192kHz XLR analog	8	8
24-bit 192kHz XLR main outputs	-	stereo
Headphone output	-	stereo
ADAT optical digital (at 44.1 or 48kHz)	8	8
AES/EBU 24-bit 96kHz digital	stereo	stereo
Total	18	22

All inputs and outputs are discrete. In other words, using the main outs does not “steal” an output pair from the bank of eight XLR analog outputs. The same is true for the headphone outs.

The ADAT optical ports provide 4 channels of I/O at 88.2 or 96kHz. They are disabled at the 4x sample rates (176.4 and 192kHz).

The headphone output can operate as an independent output pair, or it can mirror any other 896HD output pair, such as the main outs.

Analog

All 10 analog inputs are equipped with 24-bit 192kHz, 64x oversampling A/D converters. All 12 analog outputs have 24-bit 128x oversampling D/A converters. All audio is carried to the computer in a 24-bit data stream. Each output can be individually switched between either +4 or -10dB operation. Each input can be individually set to one of three input levels: MIC (feeds the mic preamp and

includes front-panel trim and switchable 48V phantom power), LINE (for -10dB inputs with front-panel trim) and +4/FIXED (for +4 “hot” inputs for which no gain adjustment in the 896HD is desired).

Mic preamps

All eight analog inputs are equipped with a mic preamp on a Neutrik™ combo-style connector that accepts either an XLR or quarter-inch plug. Defeatable 48V phantom power is supplied by a front panel switch. In addition, each input has its own trim knob, which provides a trim range of approximately 40dB.

Main Outs

The main outs are equipped with 24-bit 192kHz 128x oversampling D/A converters and serve as independent outputs for the computer or for the 896HD’s on-board CueMix DSP mixes. The main out volume can be controlled with the front panel volume knob (when the switch is in the *Main Outs + Phones* position).

Optical

The 896HD optical jacks support the industry standard ADAT optical format, which provides eight channels of 24-bit digital audio at either 44.1 or 48 kHz, and four channels at 88.2 or 96kHz.

AES/EBU with sample rate conversion

The 896HD rear panel provides a standard AES/EBU digital input and output that supports digital I/O at 44.1, 48, 88.2 and 96 kHz. In addition, input or output can be sample-rate converted to any of these sample rates in situations that call for a different rate than the 896HD’s global sample rate. The AES/EBU jacks are disabled at the 4x sample rates (176.4 and 192kHz).

ADAT sync: sample-accurate synchronization

The 896HD’s standard 9-pin ADAT SYNC IN connector provides sample-accurate synchronization with all Alesis ADAT tape decks connected

to the system—or any device that supports the ADAT sync format. For example, if you digitally transfer a single track of material from an ADAT via light pipe into audio workstation software on the computer, and then transfer the track back to the ADAT, it will be recorded exactly at its original location, down to the sample.

Word clock

The 896HD provides standard word clock that can slave to any supported sample rate. In addition, word clock can resolve to and generate “high” and “low” sample rates. For example, if the 896HD global sample rate is set to 96 kHz, the word clock input can resolve to a “low” rate of 48 kHz.

Similarly, when the 896HD is operating at 192 kHz, the MOTU FireWire Audio Console lets you choose a word clock output rate of 48 kHz (the *Force 44.1/48kHz* setting).

Punch in/out

The quarter-inch Punch in/out jack accepts a standard foot switch. When you push the foot switch, the 896HD triggers a programmable keystroke on the computer keyboard. For example, with MOTU’s Digital Performer audio sequencer software, the foot switch triggers the 3 key on the numeric keypad, which toggles recording in Digital Performer. Therefore, pressing the foot switch is the same as pressing the 3 key. The 896HD Control Panel software lets you program any keystroke you wish.

1394 FireWire

The two 1394 FireWire jacks accept a standard IEEE 1394 FireWire cable to connect the 896HD to a FireWire-equipped Macintosh or Windows computer. The second jack can be used to daisy chain multiple interfaces — up to four MOTU FireWire interfaces — on a single FireWire bus. It can also be used to connect other FireWire devices without the need for a FireWire hub.

THE 896HD FRONT PANEL

Headphone output and main volume control

The MOTU 896HD front panel includes a quarter-inch stereo headphone output jack and volume knob. From the factory, the headphone output matches the main stereo outs. An accompanying switch allows you to control the volume of the phones only (down) or both the phones and the main outs (up). The headphone output can also operate as its own independent output pair, or it can mirror any other 896HD output pair, such as the AES/EBU output. Use the MOTU FireWire Console to choose the desired headphone output.

CueMix™ DSP no-latency monitoring

The MOTU 896HD provides CueMix™ DSP no-latency monitoring, which can mix all inputs to any output pair. At samples rates up to 96kHz, four such mixes can be independently programmed and simultaneously operated. At the 4x sample rates (176.4 or 192kHz), two CueMix DSP mixes are supported.

Input trim knobs and phantom power switch

The front-panel input trim knobs provide independent trim for the eight analog inputs. The phantom power switch for each input provides 48V phantom power. Up is on; down is off.

Metering

The front panel of the MOTU 896HD displays two eight-channel banks of 10-segment ladder LEDs. The left-hand bank always shows the eight analog inputs. The right-hand bank shows one of three different banks, which you can specify in the MOTU FireWire Audio Console software: Analog out, ADAT input, or ADAT output. A status LED to the left shows which bank you are currently viewing. These two banks also provide metering for the four CueMix DSP monitor mixes (when the front-panel MONITOR LEVEL knob is pressed and/or turned).

The 896HD front panel also displays stereo meters for the main analog outs and AES/EBU. The AES/EBU meters can display either input or output as specified in the MOTU FireWire Audio Console software. A status LED to the right shows whether you are viewing input or output.

16-BIT AND 24-BIT RECORDING

The 896HD system handles all data with a 24-bit signal path, regardless of the I/O format. You can record and play back 16-bit or 24-bit audio files at any supported sample rate via any of the 896HD's analog or digital inputs and outputs. 24-bit audio files can be recorded with any compatible host application that supports 24-bit recording.

HOST AUDIO SOFTWARE

The 896HD system ships with a standard WDM Windows driver that allows you to record, edit, play back and mix your 896HD projects using your

favorite Windows audio software. A standard GSIF driver is also included for operation with Tascam GigaStudio.

The 896HD also includes a Windows ASIO driver for multi-channel compatibility with any audio application that supports ASIO drivers.

A COMPUTER-BASED SYSTEM

Regardless of what software you use with the 896HD, the host computer determines the number of tracks the software can record and play simultaneously, as well as the amount of real-time effects processing you can apply to your mix. A faster computer with more RAM and faster hard drives will allow more simultaneous tracks and real-time effects than a slower computer with less RAM and slower hard drives. Today's fastest computers can typically play as many as 72 tracks or more.

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

PACKING LIST

The 896HD ships with the items listed below. If any of these items are not present in your 896HD box when you first open it, please immediately contact your dealer or MOTU.

- One 896HD I/O rack unit
- One 1394 “FireWire” cable
- Power cord
- One 896HD Mac/Windows manual
- One AudioDesk OS 9/OS X “Flipbook” Manual
- One cross-platform CD-ROM
- Product registration card

WINDOWS SYSTEM REQUIREMENTS

The 896HD system requires the following Windows system:

- A 500 MHz Pentium-based PC compatible or faster equipped with at least one FireWire port
- A Pentium III processor or faster is recommended
- At least 128MB (megabytes) of RAM (64MB or more is recommended)
- Windows Me, 2000 or XP
- A large hard drive (preferably at least 20GB)

PLEASE REGISTER TODAY!

Please send in the registration card included with your 896HD system. As a registered user, you will be eligible to receive on-line technical support email and announcements about product enhancements as soon as they become available. Only registered users receive these special update notices, so please, complete and mail this registration card!

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

IMPORTANT! Run the 896HD Software Installer First

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 The MOTU FireWire audio WDM driver 16
 MOTU GSIF FireWire driver 16
 CueMix Console 16

INSTALL THE 896HD SOFTWARE FIRST!

Before you connect the 896HD audio interface to your computer and turn it on, insert the 896HD software CD and run the 896HD Software Installer. This ensures that all the 896HD components are properly installed in your system.

If Windows asks you to locate the drivers

If you’ve already connected the 896HD to your computer and switched it on, Windows probably issued an alert notifying you that the 896HD requires drivers, followed by another window asking you to locate the drivers on disk. If this happens:

- 1 Cancel the driver search.
- 2 Switch off the 896HD.
- 3 Run the 896HD Software Installer as instructed in the next section.

INSTALLING THE 896HD SOFTWARE

To install the 896HD software, insert the MOTU Audio CD-ROM and follow the directions it gives you on your computer screen. The 896HD ships with the following software and drivers for Windows Me, 2000 and XP:

Software component	Purpose
MOTU FireWire Audio Console	Provides access to all of the settings in the 896HD hardware.
ASIO MOTU FireWire Audio Driver	Allows Cubase or other ASIO-compliant software to do multi-channel input and output with the 896HD.
MOTU FireWire WDM Driver	Allows any WDM-driver compatible audio software to do multichannel input and output with the 896HD.
MOTU FireWire GSIF Driver	Allows you to use your 896HD with Tascam GigaStudio.
FireWire CueMix Console	Gives you complete control over the 896HD’s CueMix DSP feature, which provides no-latency monitoring of live inputs.

MOTU FIREWIRE AUDIO CONSOLE

The MOTU FireWire Audio Console (available in the Start menu) gives you access to all of the settings in the 896HD, such as the clock source and sample rate. For complete details, see chapter 5, “MOTU FireWire Audio Console” (page 35).

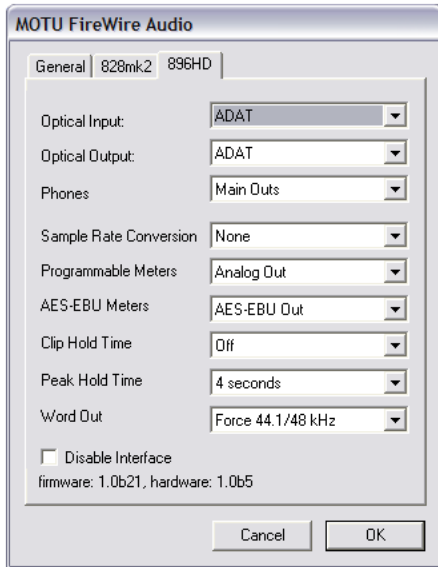


Figure 3-1: The MOTU FireWire Audio Console gives you access to all of the settings in the 896HD hardware.

ASIO MOTU FIREWIRE AUDIO DRIVER

ASIO stands for *Audio Streaming Input and Output*. The ASIO MOTU FireWire driver allows 896HD to provide multi-channel input and output for Steinberg’s Cubase VST software, or any other audio application that supports ASIO drivers.

The ASIO MOTU FireWire driver is installed by the 896HD Software Installer and properly registered with Windows, so you don’t need to be concerned about its installation or location.

For details about using Cubase VST with the 896HD, see chapter 6, “Cubase, Nuendo and Other ASIO Software” (page 43).

THE MOTU FIREWIRE AUDIO WDM DRIVER

The MOTU FireWire Audio WDM driver provides standard multi-channel input and output for audio applications running under Windows Me, 2000 or XP. See chapter 7, “Sonar and other WDM Software” (page 53) for details.

The MOTU FireWire Audio installer CD installs the MOTU FireWire Audio WDM for you.

MOTU GSIF FIREWIRE DRIVER

The MOTU GSIF FireWire Driver allows you to access the 896HD as an audio interface for Tascam GigaStudio and related products. The GSIF driver features low-latency multi-channel performance.

CUEMIX CONSOLE

This program provides a mixing console that gives you control over the 896HD’s no-latency CueMix DSP features. For details, see chapter 9, “CueMix Console” (page 65).

CHAPTER 4 Installing the 896HD Hardware

OVERVIEW

Here's an overview for installing the 896HD:

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CONNECT THE 896HD INTERFACE

- 1 Plug one end of the 896HD FireWire cable (included) into the FireWire socket on the computer as shown below in Figure 4-1.
- 2 Plug the other end of the FireWire cable into the 896HD I/O as shown below in Figure 4-1.



Figure 4-1: Connecting the 896HD to the computer.

CONNECT AUDIO INPUTS AND OUTPUTS

The 896HD audio interface has the following audio input and output connectors:

- 8 XLR analog outputs
- 8 Neutrik™ XLR/quarter-inch analog inputs
- 2 XLR main outs
- AES/EBU input/output
- ADAT optical input/output

Analog inputs

The 896HD analog inputs are Neutrik combo connectors that accept either a male XLR plug or a quarter-inch plug. You can use either type of plug, regardless of whether it is a mic, synth, or whatever. Set the 3-way input level switch as follows:

For a microphone or unamplified instrument pickup, set the rear-panel 3-way switch to MIC, plug in your mic (XLR or quarter-inch plug), flip on 48V phantom power (if necessary) and use the trim knob as needed to adjust the level.

For -10dB (unbalanced) inputs (like synths) or +4 (balanced) signals that may need to be boosted, set the rear panel 3-way switch to LINE and use the trim knob to adjust the level. You can use either an XLR or quarter-inch plug. (Note: the complete trim range, from all the way down with the MIC setting to all the way up with the LINE setting is around 55dB total. The MIC setting provides a 40dB range and the LINE setting offers a 30dB range with some overlap between them.)

If you have +4 inputs for which you'd like to maintain unity gain, set the 3-way switch to +4/FIXED. Use either an XLR or quarter-inch plug.

Analog outputs

Connect an XLR cable and set the desired output level with the 2-way level switch (+4 or -10dB).

ADAT optical

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

AES/EBU

Connect standard AES/EBU input and output. 2x sample rates (88.2 & 96 kHz) are supported; 4x samples rates (176.4 or 192kHz) are not supported.

Main outs

When the 896HD is operating at any sample rate up to 96kHz, the main outputs serve as a separate, independent output pair. The main out volume can be controlled by the main volume knob on the front panel (when the switch is up). In a typical studio, the main outs are intended for a pair of monitors. However, if you are using the 896HD in other ways, such as in a live performance situation, you could use the main outs for stage monitors or for some other purpose.

Using an external mixer

The 896HD can be used with or without a mixer, as shown on the following pages. In Figure 4-3 (no mixer), all mixing and effects processing occurs in the audio software running on the computer. If you'd like to use external mixing, see "Using the 896HD with a mixer" on page 20.

Figure 4-2: the 896HD rear panel.



CONNECT A FOOT SWITCH

If you would like to use a foot switch with your 896HD, connect it to the PUNCH IN/OUT jack. See “Quick Reference: MOTU FireWire Audio Console” on page 7 for information about how to program the foot switch to trigger any computer keystroke you wish.

A TYPICAL 896HD SETUP (NO MIXER)

Here is a typical 896HD studio setup. This rig can be operated without an external mixer. All mixing and processing can be done in the computer with audio software. During recording, you can use the 896HD’s CueMix™ DSP no-latency monitoring to listen to what you are recording via the main outs, headphone outs, or any other output pair. You can control monitoring from the included CueMix Console software.

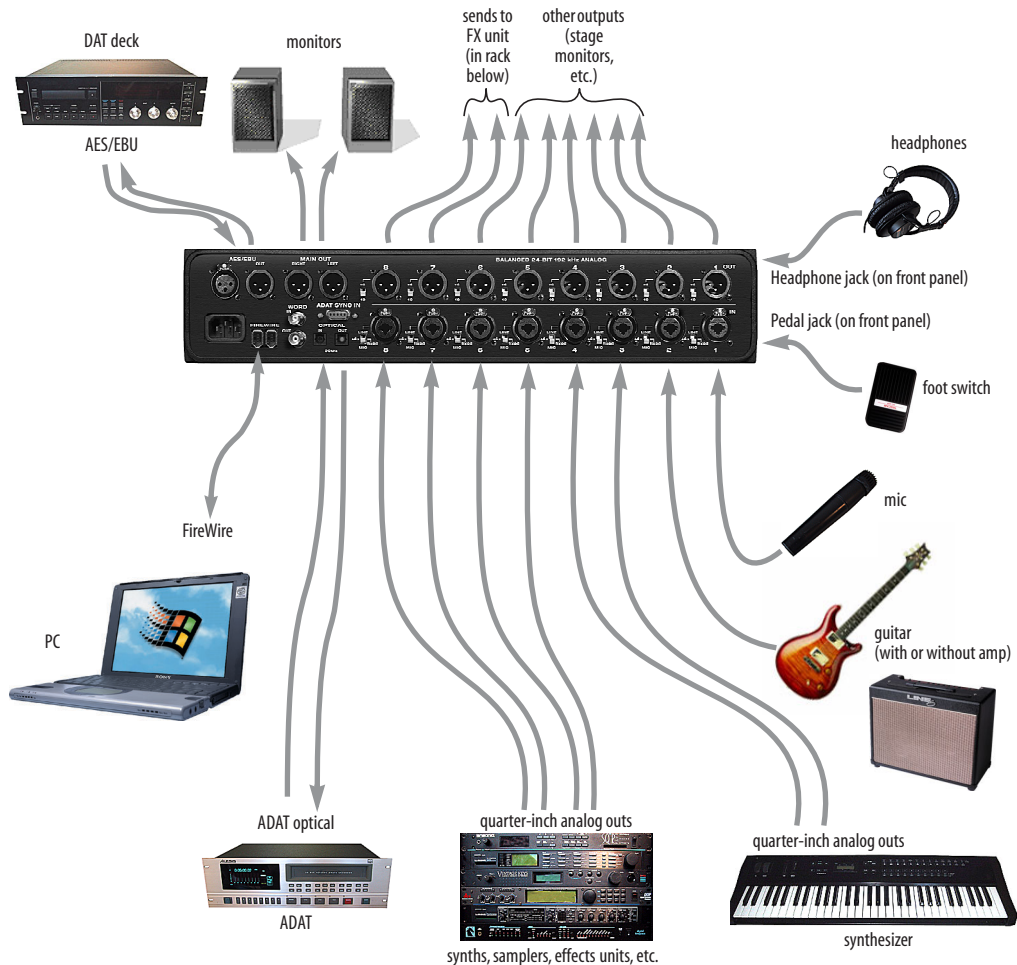


Figure 4-3: A typical 896HD studio setup.

USING THE 896HD WITH A MIXER

While there are many ways to use the 896HD with an external mixer, typically the 896HD serves as a multi-channel “pipeline” between the mixer and the computer. If your mixer is analog, connect the analog section of the 896HD to your mixer. If your mixer is digital, and it has ADAT optical I/O, you can connect them optically as shown below in Figure 4-4. Add more 896HD’s for additional

banks of eight-channel I/O. The 896HD’s available analog and AES/EBU inputs and outputs can serve as an extension to the mixer I/O, but then you will probably find yourself mixing in two places: the mixer and the computer. A word of advice: if you would like to use the 896HD with an external mixer, use the mixer for mixing. Trying to mix large multitrack projects in two places can become very cumbersome very quickly.

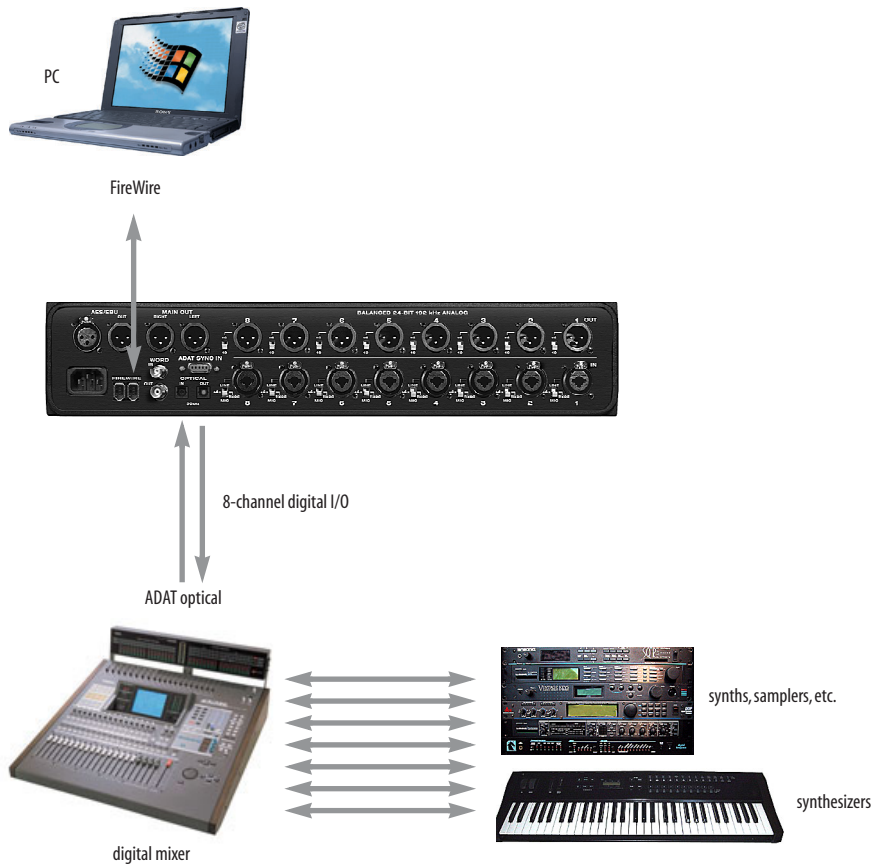


Figure 4-4: Using the 896HD with a digital mixer.

MAKING SYNC CONNECTIONS

If you connect devices digitally to the 896HD, or if you need to synchronize the 896HD with an outside time reference such as SMPTE time code, you must pay careful attention to the synchronization connections and clock source issues discussed in the next few sections.

Do you need to synchronize the 896HD?

If you will be using only the 896HD's analog inputs and outputs (and none of its digital I/O), and you have no plans to synchronize your 896HD system to SMPTE time code or other external clock source, you don't need to make any sync connections. You can skip this section and proceed to chapter 6, "MOTU FireWire Control Panel (Mac OS 9)" (page 45) where you'll open the MOTU FireWire Audio Console and set the *Clock Source* setting to *Internal* as shown below. For details, see chapter 6, "MOTU FireWire Control Panel (Mac OS 9)" (page 45).

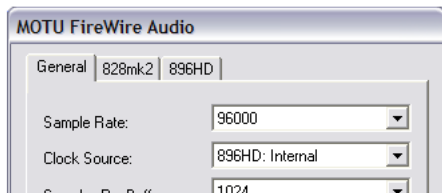


Figure 4-5: You can run the 896HD under its own internal clock when it has no digital audio connections and you are not synchronizing the 896HD system to an external time reference such as SMPTE time code.

Situations that require synchronization

There are three general cases in which you will need to resolve the 896HD with other devices:

- Synchronizing the 896HD with other digital audio devices so that their digital audio clocks are *phase-locked* (as shown in Figure 4-6)
- Slaving the 896HD system to SMPTE time code from a video deck, analog multi-track, etc.
- Both of the above

Synchronization is critical for clean digital I/O

Synchronization is critical in any audio system, but it is especially important when you are transferring audio between digital audio devices. Your success in using the 896HD's digital I/O features depends almost entirely on proper synchronization. The following sections guide you through several recommended scenarios.

Be sure to choose a digital audio clock master

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

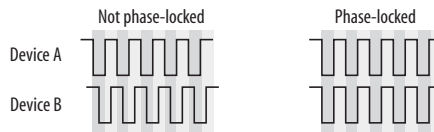


Figure 4-6: When transferring audio, two devices must have phase-locked audio clocks to prevent clicks, pops or other artifacts.

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

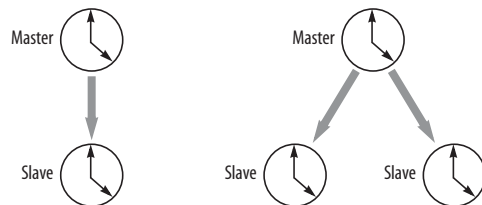


Figure 4-7: To keep the 896HD phased-locked with other digital audio devices connected to it, choose a clock master.

Also remember that audio phase lock can be achieved independently of time code (location). For example, one device can be the time code master while another is the audio clock master. But only one device can be the audio clock master. If you set things up with this rule in mind, you'll have trouble-free audio transfers with the 896HD.

DO YOU NEED A SYNCHRONIZER?

Whether or not you'll need a synchronizer depends on your gear and what you will be doing with your 896HD system. The following pages give you specific information about common sync scenarios. At least one of them will likely apply to you. Here are some general considerations to help you determine if you need (or want) a synchronizer for your 896HD system.

You don't need a synchronizer if...

As explained earlier, the 896HD's digital audio clock must be phase-locked (synchronized) with other connected digital audio devices to achieve clean digital transfers between them. Can this be accomplished without an additional digital audio synchronizer? It depends on the nature of the other devices, and what you want to do with them. You don't need a synchronizer if the device has a way of locking itself directly to the 896HD's clock (via ADAT lightpipe, AES/EBU or word clock), AND if the device carries no sense of location in time. A digital mixer is a good example: it can slave to its ADAT lightpipe connection from the 896HD, and it has no sense of time; it just passes audio through for mixing.

A stand-alone digital recorder, on the other hand, does have a sense of location in time, either via SMPTE time code or via its own sample address. For example, if you want to fly tracks back and forth between your computer and an Alesis hard disk recorder while maintaining the audio's position in time, the ADAT Sync port on the 896HD lets you do so without a separate synchronizer — and with sample-accurate precision, as long as you're using sample-accurate software. Just connect the 896HD directly to the Alesis recorder (or other ADAT Sync-compatible device) as discussed in “Sample-accurate ADAT sync with no synchronizer” on page 25. But if you also want transport control over the entire rig (including the hard disk recorder) from your audio

software, you'll need a MIDI Machine Control-compatible synchronizer such as MOTU's MIDI Timepiece AV, as discussed in “Sample-accurate sync” on page 23. If you are simply using a stand-alone recorder as a way to capture live tracks that you then transfer in one pass into the computer, no synchronizer is required because the tracks will remain in perfect phase lock with each other as you transfer them together. You can simply slave the stand-alone recorder to the optical output from the 896HD as explained in “Syncing optical devices” on page 27.

Transport control from your computer

If you have stand-alone digital recorders connected to the 896HD, and they support ADAT Sync, your audio software — if it supports MIDI Machine Control (MMC) — allows you to control the transports of everything from your computer. Most advanced audio programs support MMC. To do this, you'll also need an MMC-compatible ADAT synchronizer such as a MOTU MIDI Timepiece AV or Digital Timepiece. Synchronizers like these allow you to play, stop, rewind and locate all of your tape decks using the transport controls in the audio software. If your audio software supports sample-accurate sync, you can do so with sample-accurate precision. The following pages show you how to achieve MMC control, where possible.

Continuous sync to SMPTE / MTC

If you need to synchronize the 896HD (and your audio software) to SMPTE time code, this requires a dedicated synchronizer, which continuously resolves the 896HD to SMPTE time code, while simultaneously resolving your audio software to MIDI Time Code. When the 896HD is continuously resolved, audio playback will never drift with respect to the time code. Again, the MOTU MIDI Timepiece AV and Digital Timepiece are affordable examples of this type of synchronizer. The following pages illustrate how to set up this type of synchronization with various

kinds of gear. Regardless of the specific equipment you have, you can follow the basic connections shown.

SAMPLE-ACCURATE SYNC

Your 896HD system provides you with the most advanced, accurate synchronization possible with Alesis modular digital tape decks and hard disk recorders — or any device that supports sample-accurate ADAT sync. Figure 4-8 below shows a few best-case scenarios. Below is a brief explanation of the benefits you achieve with these setups.

Sample accurate locating

With *sample accurate locating*, when you transfer audio between any other sample-accurate host software and a sample-accurate recorder, the audio will not drift in time — even by as little as one sample. This is the tightest possible synchronization between digital audio devices. The timing in your audio will not be affected in any way by the process of transferring it between the 896HD and the recorder.

Is your audio software sample-accurate?

Sample-accurate locating is only possible with software that supports this feature through the 896HD's ASIO Version 2 driver.

Transport control from your computer

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

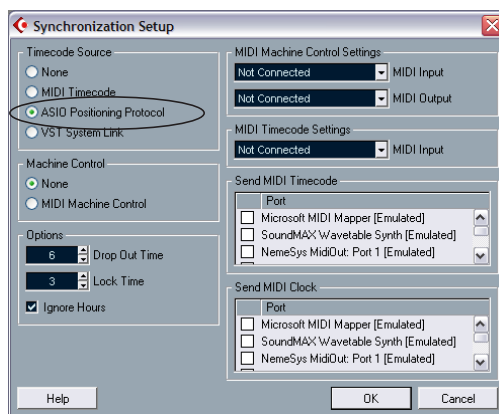


Figure 4-9: Cubase SX and other ASIO 2.0-compatible programs support sample-accurate transfers with ADAT Sync compatible digital tape decks and modular hard disk recorders.

Sync format	Software	Synchronizer	Sample accurate locating	Transport control from computer	Continuous sync to SMPTE / MTC
ADAT	Cubase or other ASIO 2.0-compatible app	MIDI Timepiece AV or Digital Timepiece	Yes	Yes	Yes
ADAT	Cubase or other ASIO 2.0-compatible app	BRC (or any MMC capable ADAT synchronizer)	Yes	Yes	Yes
ADAT	Cubase or other ASIO 2.0-compatible app	None	Yes	No	No

Figure 4-8: These recommended combinations of hardware and software offer the tightest sync possible between the 896HD and digital audio recorders in the form of sample-accurate locating between the software and the tape decks. Sample accurate locating is possible even without a MIDI Timepiece AV or Digital Timepiece, although you give up transport control from the computer.

SAMPLE-ACCURATE ADAT SYNC

The 896HD can achieve sample-accurate sync with ADATs, Alesis hard disk recorders or any ADAT Sync-compatible devices. Sample-accurate software is required, such as Cubase SX or other ASIO 2.0-compatible software that also supports sample-accurate sync. Connect the 896HD to the end of the ADAT Sync chain and make the software settings shown below in Figure 4-10. If you will be using the stand-alone recorder for its analog inputs and outputs only (you won't be doing any recording with it), treat it as an 'optical' device. See "Syncing optical devices" on page 27.

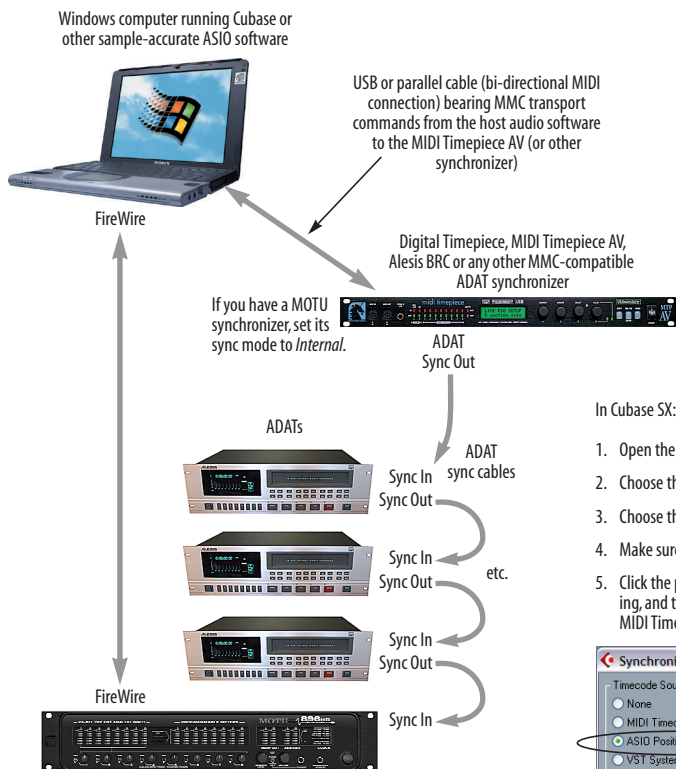


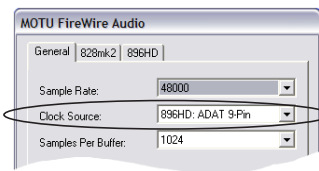
Figure 4-10: Connections for sample-accurate ADAT sync.

Use this setup if you have:

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

This setup provides:

- ✓ Sample-accurate locating between all ADAT SYNC-compatible devices, the 896HD and your software (Cubase or other sample-accurate software).
- ✓ With a Digital Timepiece, this setup provides sample-accurate locating across all devices: ADAT, Tascam and the 896HD.
- ✓ Transport control of everything from the computer, OR continuous sync to SMPTE time code and other sync sources (the other source is the transport master in this case).

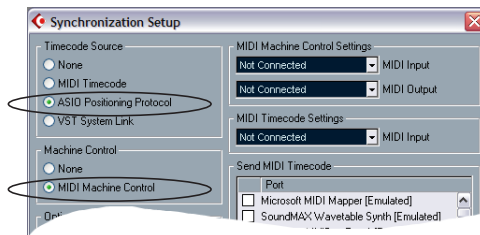


To set the 896HD hardware clock source for sample-accurate sync:

1. Run the MOTU FireWire Audio Console.
2. Choose ADAT 9-pin from the Clock Source menu as shown above.
3. Make sure the Sample Rate setting matches the recorder and synchronizer.

In Cubase SX:

1. Open the Sync Setup window (Transport menu).
2. Choose the *ASIO Positioning Protocol* option shown below.
3. Choose the *MIDI Machine Control* option shown below.
4. Make sure that *Sync Online* is checked in the Transport menu.
5. Click the play or record button. Cubase will then start playing or recording, and the recorder will begin to roll, too, after being triggered by the MIDI Timepiece AV (or other synchronizer).



SAMPLE-ACCURATE ADAT SYNC WITH NO SYNCHRONIZER

Even if you don't have an ADAT synchronizer, you can achieve sample-accurate sync between ADAT-sync compatible devices, an 896HD, and any sample-accurate software. Just connect the 896HD to the end of the ADAT sync chain as shown below. You don't get transport control from your computer, nor can you slave the system to SMPTE time code. Instead, you have to play, stop, rewind and cue the system from the transports on your recorder. If you're using the recorder as an additional source of analog inputs and outputs only (not for recording), see "Syncing optical devices" on page 27.

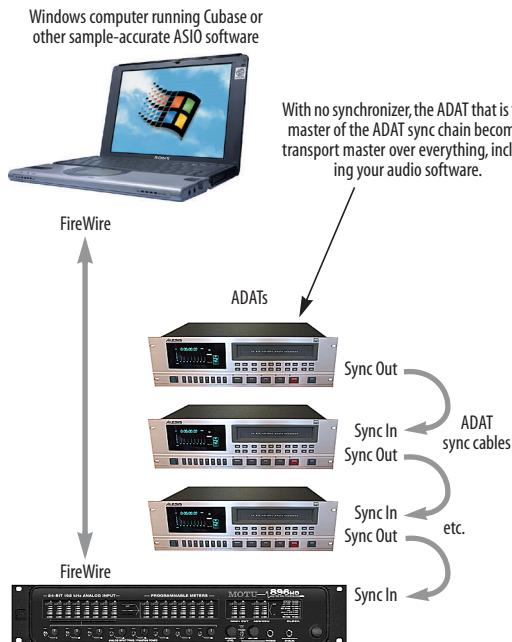


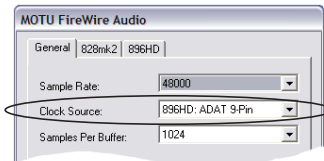
Figure 4-10: Sample-accurate sync without an ADAT synchronizer.

Use this setup if you have:

- ✓ ADATs, Alesis hard disk recorders or any ADAT SYNC compatible device(s).
- ✗ No ADAT synchronizer.
- ✓ Host software that supports sample-accurate sync.

This setup provides:

- ✓ Sample-accurate locating between all ADAT SYNC-compatible devices, the 896HD and your software (Cubase or other sample-accurate software).
- ✗ No transport control of everything from the computer.
- ✗ No sync to SMPTE time code or other sync sources.

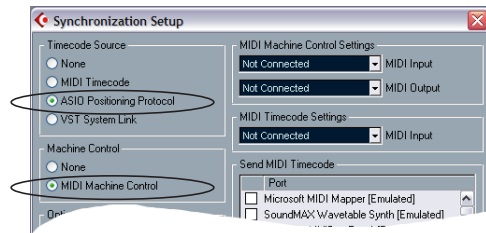


To set the 896HD hardware clock source for sample-accurate sync:

1. Run the MOTU FireWire Audio Console.
2. Choose ADAT 9-pin from the Clock Source menu as shown above.
3. Make sure the Sample Rate setting matches the recorder and synchronizer.

In Cubase SX:

1. Open the Sync Setup window (Transport menu).
2. Choose the *ASIO Positioning Protocol* option shown below.
3. Choose the *MIDI Machine Control* option shown below.
4. Make sure that *Sync Online* is checked in the Transport menu.
5. Click the play or record button. Cubase will then start playing or recording, and the recorder will begin to roll, too, after being triggered by the MIDI Timepiece AV (or other synchronizer).



SYNCING TO VIDEO AND/OR SMPTE TIME CODE

To synchronize (continuously resolve) the 896HD with SMPTE time code, word clock, video or blackburst, you will need a MOTU Digital Timepiece, MIDI Timepiece AV or any other universal synchronizer equipped with word clock. The synchronizer resolves continuously to the chosen time base, and the 896HD slaves to the synchronizer via word clock. In addition, the audio software running on the computer slaves to MIDI Time Code generated by the synchronizer, as shown below in Figure 4-11. How accurate will transfers be between your audio software and other audio devices? As good as the resolution of MIDI time code, which — at 30 fps — provides quarter frame resolution of 120th of a second (367 samples at 44.1 KHz). But if you are running your

synchronizer under its own internal clock (triggering it via MMC from your software), you will probably get even tighter timing than that — perhaps as good as ± 50 samples.

Use this setup if you have:

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

This setup provides:

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

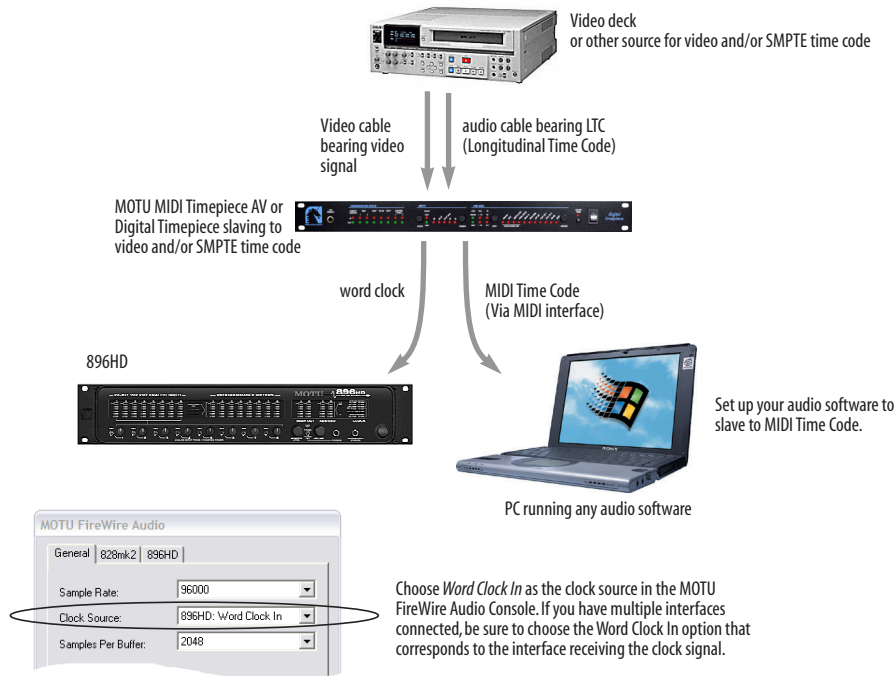


Figure 4-11: Resolving the 896HD to an external time base, such as SMPTE time code, word clock, or video. In this example, an S-VHS video deck is supplying SMPTE time code (address) and video (as the time base). For examples of other sources, consult the MIDI Timepiece AV manual (or other synchronizer).

SYNCING OPTICAL DEVICES

The word *optical* is our short-hand way of referring to any device that connects to the 896HD via an optical cable. But we make a further distinction: an optical device is also one that doesn't care about sample location. An example is a digital mixer. Since a digital mixer is not a recording device, it has no sense of sample location like an ADAT does. An ADAT can cue to a specific sample number (e.g. sample number 43,478, 103) — as can any device that supports ADAT sync, but most digital mixers simply mix and process audio digitally, with no sense of a specific sample location. There are many other devices that fall into this category, including digital effects processors, synthesizers, A/D converters, and many more.

For ADATs or other devices that support ADAT sync, synchronize them with the 896HD as described in the previous sections of this chapter.

For *optical* devices, such as digital mixers, all you have to do is make sure that their digital audio clock is phase-locked (in sync with) the 896HD. There are three ways to do this:

- Slave the optical device to the 896HD
- Slave the 896HD to the optical device
- Slave both the optical device and the 896HD to a third master clock (such as a Digital Timepiece or MIDI Timepiece AV synchronizer)

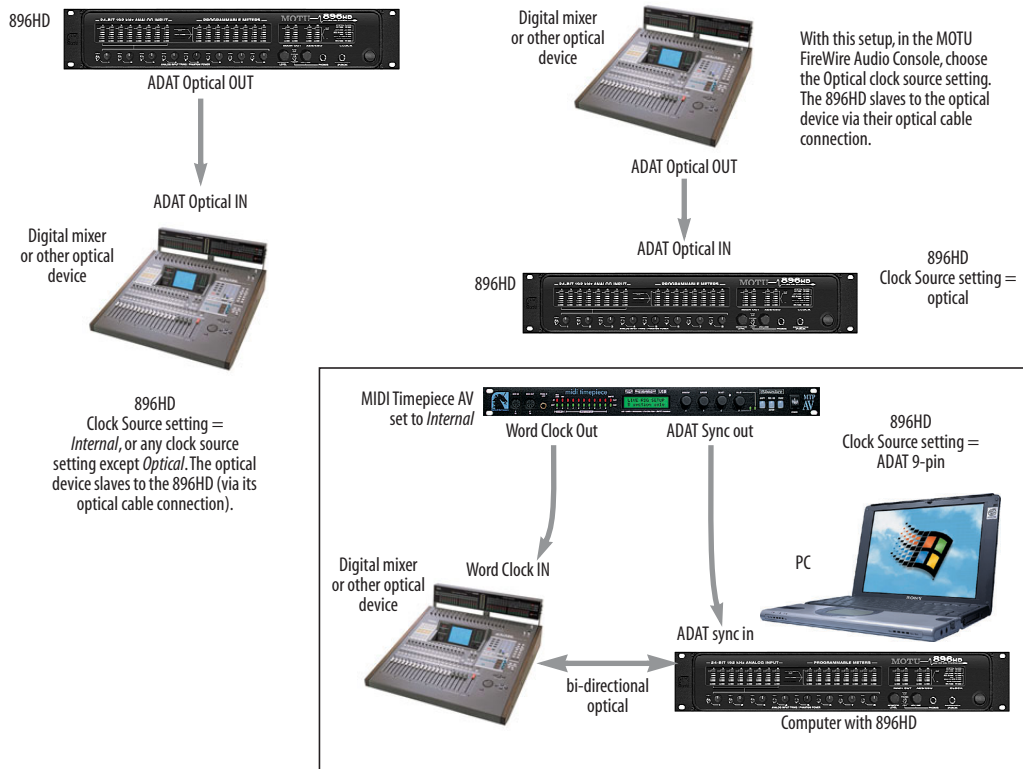


Figure 4-12: Three setups for synchronizing an optical device with the 896HD. You can slave the optical device to the 896HD or vice versa with their optical connections. For more elaborate setups, you can slave both to a digital audio synchronizer like the Digital Timepiece. Don't use any of these setups for an ADAT or other optical device that records. Instead, see "Sample-accurate ADAT sync" on page 24.

SYNCING AES/EBU DEVICES

If you would like to transfer stereo audio digitally between the 896HD and another device that has AES/EBU I/O, connect it to the 896HD's AES/EBU jacks with balanced, AES/EBU grade audio cables.

AES/EBU clock and sample rate conversion

The 896HD AES/EBU section is equipped with a real-time sample rate converter that can be used for either input or output. This feature provides a great deal of flexibility in making digital transfers. For example, you can:

- Transfer digital audio into the 896HD at a sample rate that is completely different than the 896HD system clock rate.

- Transfer digital audio into the 896HD without the need for any external synchronization arrangements.

- Transfer digital audio out of the 896HD at double or half the 896HD system clock rate.

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

Digital audio phase lock

Without sample rate conversion, when you transfer digital audio between two devices, their audio clocks must be in phase with one another — or *phase-locked* — as demonstrated below in Figure 4-14. Otherwise, you'll hear clicks, pops, and distortion in the audio, or perhaps no audio at all.

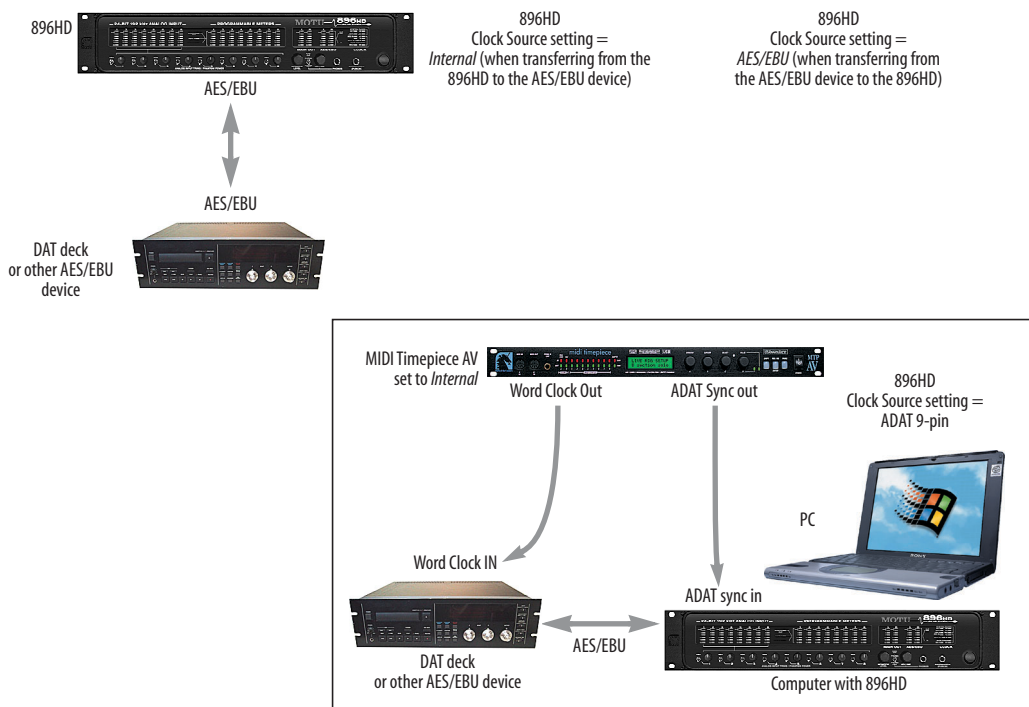


Figure 4-13: Two setups for synchronizing an AES/EBU device with the 896HD. In the top diagram, sync is achieved via the AES/EBU connection itself. In this case, you have to choose AES/EBU as the 896HD's clock source when recording from the AES/EBU device. If you don't want to have to worry about switching the Clock Source setting depending on the direction of the AES/EBU transfer, you can slave the AES/EBU device to word clock from the 896HD or vice versa (not shown). The Word Clock connection maintains sync, regardless of the direction of the transfer.

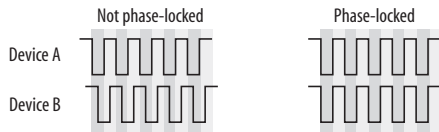


Figure 4-14: When transferring audio without sample rate conversion, two devices must have phase-locked audio clocks to prevent clicks, pops or other artifacts.

Without sample rate conversion, there are two ways to achieve phase lock: slave one device to the other, or slave both devices to a third master clock. If you have three or more digital audio devices, you need to slave them all to a single master audio clock.

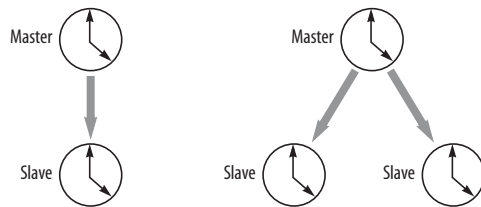


Figure 4-15: Without sample rate conversion, you need to choose a clock master to which all other devices slave. Each slaved device remains continuously resolved to the master, meaning that there will be no drift over time.

Audio phase lock as shown above in Figure 4-15 can be achieved independently of time code (location). For example, one device can be the time code master while another is the audio clock master. But only one device can be the audio clock master.

Another benefit of direct master/slave clocking (without sample rate conversion) is that each slaved device remains continuously resolved to the master, which means that there will be no gradual drift over time. This form of synchronization is best for audio that needs to remain resolved to film, video, etc.

Sample rate conversion

With sample rate conversion (SRC), an extra level of master/slave clocking is added to the equation, as demonstrated below in Figure 4-16, which shows the clocking going on when you transfer digital audio from the 896HD (AES/EBU OUT) to

a DAT deck (AES/EBU IN) using SRC. Notice that with SRC, the DAT deck is not slaved to the 896HD's system clock. Instead, their clocks are running completely independently of one another. But also notice that the DAT deck must still slave to the sample-rate-converted output from the 896HD for a clean digital audio transfer (unless it has its own sample rate converter on its AES/EBU input).

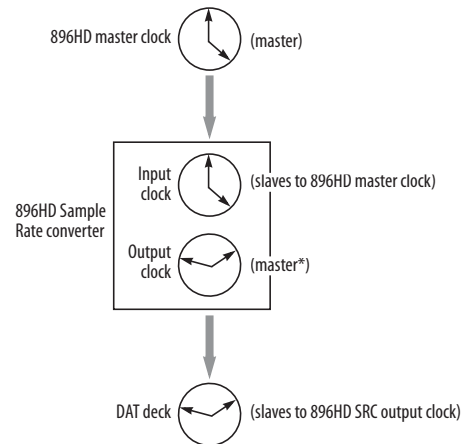


Figure 4-16: Clock relationships when sending audio from the 896HD to a DAT deck using sample rate conversion. The DAT deck needs to be slaving to its AES/EBU input. *Note: the 896HD AES/EBU output can actually be clocked from a number of different sources. In this example, it is resolved to the 896HD system clock. For details about other possible clock sources, see “Clocking scenarios for AES/EBU output” on page 30.

System clock, AES clock & rate convert settings

When you are setting up AES/EBU input and output with the 896HD, pay careful attention to the following settings in the MOTU FireWire Audio Console (see the quick reference overview on page 7):

- Clock source
- Sample rate conversion

These options are mentioned briefly in the following sections. For further details, see “Clock Source” on page 37 and “Sample Rate Conversion” on page 40.

Clocking scenarios for AES/EBU input

There are three possible clocking scenarios for the 896HD AES/EBU input:

1. Simple transfer (slave the 896HD system clock to the AES/EBU input signal — no sample rate conversion).
2. Sample rate convert the AES/EBU input.
3. Use word clock to resolve the 896HD system clock and the other AES/EBU device with each other.

These three AES/EBU input scenarios are summarized below.

	Scenario 1	Scenario 2	Scenario 3
Description	Simple transfer	Rate convert	Use word clock
896HD clock source setting	AES/EBU	Any setting except AES/EBU	Word Clock
Sample rate conversion setting	None	AES In	None
Required 896HD cable connections	AES/EBU In	AES In	AES/EBU In and Word Clock In
Are the devices continuously resolved?	Yes	No	Yes
Is the signal being sample rate converted?	No	Yes	No
Example application	Simple digital transfer into the 896HD from DAT deck or digital mixer.	Transfer from digital mixer running at a different sample rate.	Both the 896HD and other AES/EBU device are slaved to 'house' word clock.

Some example scenarios are demonstrated below.

Simple AES/EBU input transfer (no rate convert)

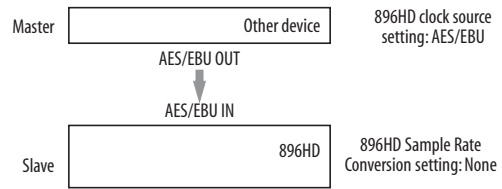


Figure 4-17: Slaving the 896HD to an AES/EBU device. For the 896HD's clock source, choose 'AES/EBU'.

AES/EBU input with rate conversion

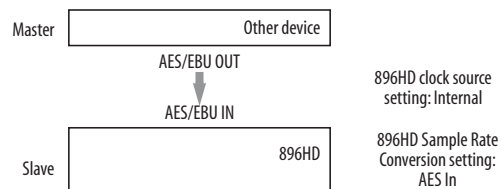


Figure 4-18: Rate-converting AES/EBU input.

AES/EBU input with word clock

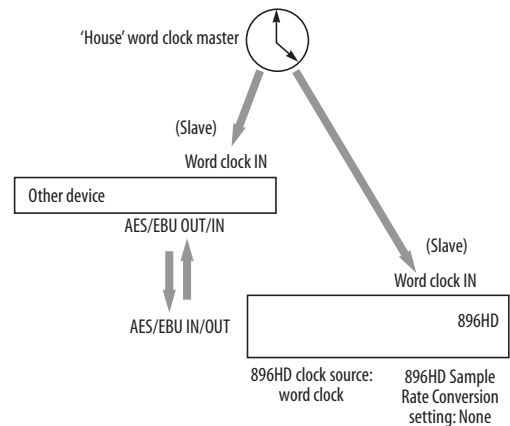


Figure 4-19: In this scenario, the 896HD and other AES/EBU device are both resolved to one another via a third master word clock source.

Clocking scenarios for AES/EBU output

The 896HD AES/EBU output can also employ sample rate conversion. The output options, shown below in Figure 4-20, are briefly summarized in the following sections. For further details, see “Sample Rate Conversion” on page 40.

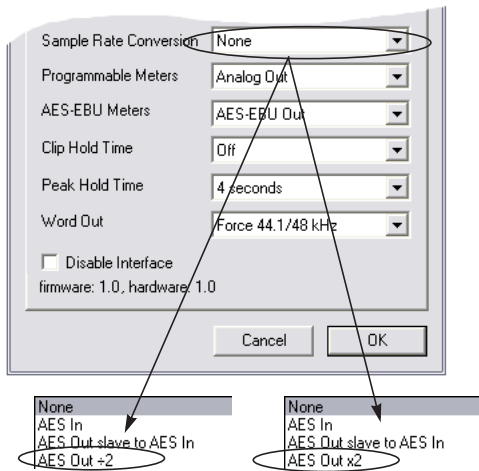


Figure 4-20: The Sample Rate Conversion option in the MOTU FireWire Audio Console gives you access to the AES/EBU output clock options. The last option is either “x2” or “÷2” depending on the system sample rate.

None

To make the AES/EBU output sample rate match the System sample rate, choose *None*. No sample rate conversion occurs when this setting is chosen.

AES Out slave to AES in

To make the AES/EBU output sample rate match the sample rate currently being received by the 896HD’s AES/EBU input, choose *AES Input*. This setting requires a connection to the 896HD’s AES/EBU input from a device that is transmitting an AES/EBU clock signal.

☛ Be careful when both the 896HD’s AES/EBU input and output are connected to the same external device: this option is likely to create a clock loop.

☛ When you are using the AES/EBU input as a clock source for sample rate conversion on the AES/EBU output, you cannot use the AES/EBU input for audio input.

AES Out x 2 / AES Out ÷ 2

Choose one of these sample rates when the desired AES/EBU output rate needs to be twice the 896HD system clock rate (when the system clock is at either 44.1 or 48 kHz) or half the system clock rate (when the system clock is at 88.2 or 96 kHz). For further details about this option, see “Sample Rate Conversion” on page 40.

SYNCING WORD CLOCK DEVICES

The 896HD word clock connectors allow you to synchronize it with a wide variety of other word clock-equipped devices.

For standard word clock sync, you need to choose an audio clock master (as explained in “Be sure to choose a digital audio clock master” on page 21). In the simplest case, you have two devices and one is the word clock master and the other is the slave as shown below in Figure 4-21 and Figure 4-22.



Figure 4-21: Slaving another digital audio device to the 896HD via word clock. For the 896HD clock source, choose any source besides word clock, as it is not advisable to chain word clock.

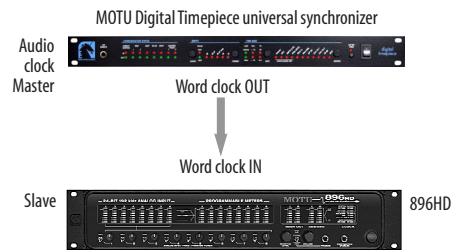


Figure 4-22: Slaving the 896HD to word clock. For the 896HD clock source, choose 'Word Clock In'.

Don't chain word clock

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

Slaving to a 2x and 1/2x word clock

The 896HD has the ability to slave to a word clock signal running at either twice or half their current clock rate. For example, the 896HD could be running at 96 kHz while slaving to a 48 kHz word clock signal from a MOTU Digital Timepiece. Similarly, the 896HD could run at 88.2 kHz and slave to 44.1 kHz word clock. Conversely, the 896HD could run at 48 kHz and slave to a 96 kHz word clock signal. In all of these cases, the front panel clock LEDs flash both sample rates to indicate that the 896HD is slaving to word clock at either twice or half its own clock rate. But if the 896HD is running at 96 kHz, it cannot slave to word clock running at 44.1 kHz.

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

- the same as the 896HD clock
- twice the 896HD clock
- half of the 896HD clock

Forcing a 1x word out rate

The 896HD can generate a word clock output signal that either matches the current system clock rate (any rate between 44.1 and 192kHz) or the corresponding 1x rate. For example, if the 896HD is operating at 192kHz, you can choose to generate a word out rate of 48kHz. For details on how to make this word clock output setting, see “Word Out” on page 41.

SYNCING LARGE SYSTEMS

If you are connecting the 896HD to a lot of other digital audio gear, get a Digital Timepiece. It can synchronize a wide variety of devices, and it offers sample accurate synchronization for devices that support it, such as ADATs. You will also be able to control everything from the transport controls of your audio software. If you have even more devices than a single Digital Timepiece can support, consider a word clock distribution device, such as the Aardvark Aard Sync™ video-to-word clock converter. Products like this offer multiple word clock outputs and an extremely low-jitter clock.

CONNECTING MULTIPLE MOTU FIREWIRE INTERFACES

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

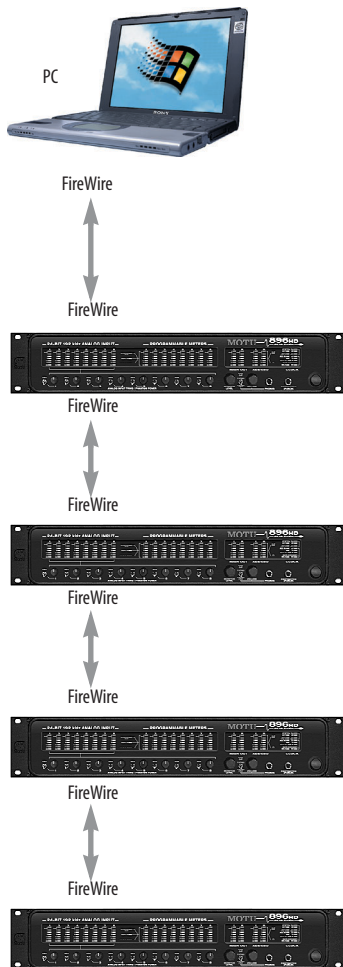


Figure 4-23: Connecting multiple 896HD's (or other MOTU FireWire audio interfaces) to a computer.

Multiple interfaces in the MOTU FireWire Audio Console

The MOTU FireWire Audio Console displays the settings for one interface at a time. To view the settings for an interface, click its tab as shown below in Figure 4-24.

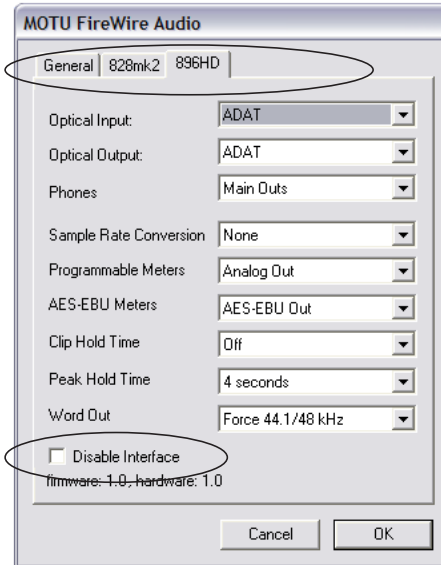


Figure 4-24: To view the settings for an interface, click its tab.

Using multiple interfaces

All connected MOTU FireWire interfaces get their clock from whatever you choose from the *Clock Source* menu in the *General* tab in the MOTU FireWire Audio Console. When you connect multiple MOTU FireWire interfaces, all of their respective sync sources are displayed in the menu as shown below in Figure 4-25.

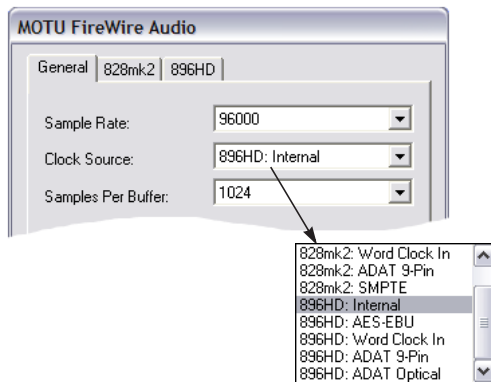


Figure 4-25: All MOTU FireWire audio interfaces get their clock from a single master sync source on any connected 896HD (or other MOTU FireWire interface). After you choose a source from this menu, the entire system, including all connected 896HDs, synchronizes to it.

Each FireWire interface in the system gets its clock from the FireWire cable connection (unless it is the master clock itself). There is no need to make word clock connections between multiple FireWire interfaces.

Connecting an 828

You can add an original MOTU 828 to the end of a FireWire daisy chain (because the 828 has only one FireWire port), or you can mix and match multiple 828's with other MOTU FireWire interfaces using a standard FireWire hub. Up to four interfaces can be combined on one FireWire bus.

Operating multiple FireWire interfaces at high sample rates

Four MOTU FireWire interfaces can operate at 44.1 or 48kHz on a single FireWire bus. At the high samples rates (88.2 or 96kHz), you can operate no more than two FireWire interfaces on a single FireWire bus.

Adding additional interfaces with a second FireWire bus

Third-party FireWire bus expansion products in the form of a cardbus (“PC card”) adaptor or PCI card allow you to add a second FireWire bus to your computer. It may be possible to add additional MOTU FireWire interfaces connected to such a third-party product, depending on the performance of the product and the performance of your host computer.

Managing the IDs of multiple interfaces

Multiple 896HD interfaces are identified by number (#1, #2, #3, etc.) Interfaces are ID'd (given a number) by the order in which they are first powered up after being connected. This information is stored in the MOTU FireWire Audio preferences file. Once ID'd, they retain the same number regardless of the order in which they are powered up. You can disable an interface at any time with the *Disable interface* option shown in Figure 4-24 on page 33. Doing so frees up the FireWire bandwidth required by the interface without turning it off. Switching off an interface accomplishes the same thing. To get the MOTU FireWire Audio Control Panel Console to forget about an interface entirely, you'll see a *Forget* button in the MOTU FireWire Audio Console. Just click the *Forget* button and the MOTU FireWire Audio Console will no longer consider the interface to be present but off line (turned off).

CHAPTER 5 MOTU FireWire Audio Console

OVERVIEW

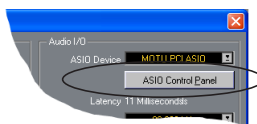
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ACCESSING THE 896HD SETTINGS

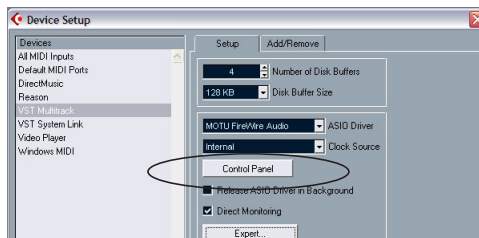
There are several ways to access the MOTU FireWire Audio Console settings:

- From the Windows Start menu, choose *Programs>MOTU>MOTU FireWire Audio Console*
- From within Cubase VST, go to the Audio menu, choose System and then click the ASIO Control Panel button. In Cubase SX, open the Devices Setup window, click the VST Multitrack device and click the Control Panel button.

Cubase VST



Cubase SX



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

General tab settings

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

896HD tab settings

The 896HD tab provides settings that apply to a specific 896HD interface. If you have several 896HD's (or other MOTU FireWire audio interfaces) connected, you'll see a separate tab for each one.

'GENERAL' TAB SETTINGS

Sample Rate

Choose the desired *Sample Rate* for recording and playback. The 896HD can operate at 44.1 (the standard rate for compact disc audio), 48, 88.2, 96, 176.4 or 192KHz. If you are operating at a sample rate between 44.1 and 96kHz, make absolutely sure that all of the devices connected digitally to the 896HD match the 896HD's sample rate. Also make sure that your Digital Timepiece, MIDI Timepiece AV or other digital audio synchronizer matches it as well. At the 4x sample rates (176.4 or 192kHz), all digital I/O on the 896HD is disabled.

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

Operation at 4x sample rates (176.4 or 192kHz)

At the 4x sample rates (176.4 or 192kHz), operation of the 896HD is restricted, due to the higher audio bandwidth demands, as follows:

- All digital I/O is disabled (there is no optical and no AES/EBU input/output).
- The 896HD provides 8 channels of analog input and 10 channels of analog output (8 XLR outputs plus stereo headphone out), simultaneously.
- CueMix DSP supports 2 independent monitor mixes (instead of 4, as with the lower sample rates). For details about CueMix DSP, see chapter 9, "CueMix Console" (page 65).
- The *Mix1* input, as described in "Mix1 1-2" on page 64, is not available.

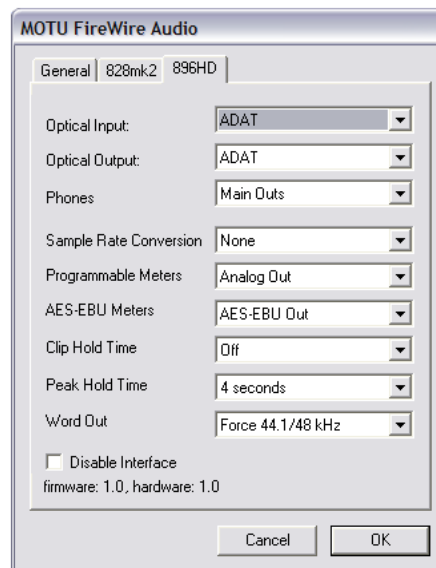
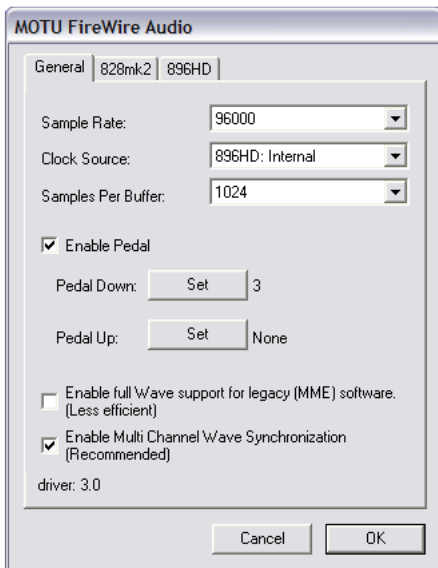


Figure 5-1: The MOTU FireWire Audio Console gives you access to all of the settings in the 896HD hardware.

- The headphone output can be assigned to any analog output pair or the *Phones* setting (as described in “Phones” on page 40). But at the 4x sample rates, the Phones output is not available from the computer. Instead, it is only available as a destination for the two CueMix DSP mixes. In other words, it can only take CueMix inputs.

- The main outs mirror the phones.

Clock Source

The *Clock Source* determines the digital audio clock that the 896HD will use as its time base. For a complete explanation of synchronization issues, see “Making sync connections” on page 21. The following sections briefly discuss each clock source setting.

Internal

Use the *Internal* setting when you want the 896HD to operate under its own digital audio clock. For example, you may be in a situation where all you are doing is playing tracks off hard disk in your digital audio software on the computer. In a situation like this, you most often don't need to reference an external clock of any kind.

Another example is transferring a mix to DAT. You can operate the 896HD system on its internal clock, and then slave the DAT deck to the 896HD via the AES/EBU connection (usually DAT decks slave to their AES/EBU input when you choose the AES/EBU input as their record source) or via the 896HD's word clock output (if your DAT deck has a word clock input).

If you would like help determining if this is the proper clock setting for your situation, see “Do you need a synchronizer?” on page 22.

With ADAT devices, however, you usually want an external digital audio synchronizer, such as the MIDI Timepiece AV or Digital Timepiece, to be the

digital clock master. In this case, you would set the 896HD clock source setting to *ADAT 9-pin*, as described below.

AES/EBU

The *AES/EBU* clock source setting refers to the AES/EBU input connector on the 896HD. This setting allows the MOTU 896HD to slave to another AES/EBU device.

Use this setting whenever you are recording input from a DAT deck or other AES/EBU device into the 896HD. It is not necessary in the opposite direction (when you are transferring from the 896HD to the DAT machine).

For further details about this setting, see “Syncing AES/EBU devices” on page 28.

Word Clock In

The *Word Clock In* setting refers to the Word Clock In BNC connector on the 896HD rear panel. Choosing this setting allows the 896HD to slave to an external word clock source, such as the word clock output from a digital mixer.

ADAT 9-pin

The *ADAT 9-pin* clock source setting refers to the ADAT digital audio synchronization format. It allows the 896HD to slave to an ADAT — or ADAT sync chain — via its ADAT sync 9-pin connector. ADAT sync also carries precise, sample location information, which allows Cubase or other sample-accurate ASIO 2.0-compatible audio software to transfer audio to and from ADAT-sync compatible recorders without drifting by as much as one sample.

Use this setting when you are using the 896HD with one or more ADAT-sync compatible recorders. Make sure the 896HD is connected to the end of the ADAT sync chain.

You should also use this setting if you have a MIDI Timepiece AV, which allows you to drive your entire system from the transport controls of your audio software.

You could also use ADAT sync to continuously resolve the 896HD to SMPTE time code, video, and word clock via a synchronizer like the MOTU MIDI Timepiece AV. Word clock can accomplish the same thing.

For further details, see “Sample-accurate ADAT sync” on page 24, “Sample-accurate ADAT sync with no synchronizer” on page 25 and “Syncing to video and/or SMPTE time code” on page 26.

ADAT optical

The *ADAT optical* clock source setting refers to the clock provided by the 896HD’s optical input, when it is connected to an ADAT optical device. This setting can be used to slave the 896HD directly to the optical input connection. Most of the time, you can set up a better operating scenario that uses one of the other synchronization options. However, there may be occasions when you have an ADAT optical compatible device that has no way of synchronizing digitally to the 896HD or an external synchronizer such as the Digital Timepiece. In this case, the *ADAT Optical* clock source setting lets you slave the 896HD to the device itself via its digital input to the 896HD.

If the *ADAT Optical* setting does not appear in the Clock Source menu, it means that the 896HD’s optical input is currently turned off. Choose the *ADAT optical* format from the Optical input menu (Figure 5-1 on page 36).

This setting is also useful if you just need to make a simple, click-free digital transfer between the 896HD and another device — where a time code reference and shared transport control are not needed — without having to set up an elaborate synchronization scenario.

For further details about this setting, see “Syncing optical devices” on page 27.

Other audio interfaces

You may see other audio interfaces in the Clock Source list, such as another MOTU FireWire interface, a MOTU PCI-324 or PCI-424 system, the Macintosh built-in audio, or perhaps even another third-party audio interface. The 896HD can resolve to these other audio devices via their CoreAudio driver. This allows you to play and record audio with your host audio software via both interfaces at the same time without their audio streams drifting apart from one another over long recording or playback passes. No external synchronization connections are required for this setting, as the two devices are entirely resolved via the software driver.

Samples Per Buffer

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

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

☛ If you don’t need to process an incoming live signal with software plug-ins, you can monitor the signal with no delay at all using CueMix Console, which routes the signal directly to your speakers via hardware. For details, see chapter 9, “CueMix Console” (page 65).

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

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

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

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

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

The *Samples Per Buffer* setting also impacts how quickly your audio software will respond when you begin playback, although not by amounts that are very noticeable. Lowering the *Samples Per Buffer*

will make your software respond faster; raising the *Samples Per Buffer* will make it a little bit slower, but barely enough to notice.

Monitoring live inputs without plug-in effects

As mentioned earlier, CueMix Console allows you to monitor dry, unprocessed live inputs with no delay at all. For complete details, see chapter 8, "Reducing Monitoring Latency" (page 59).

Enable Pedal

Check the Enable Pedal option if a foot switch is connected to the 896HD and you would like to trigger recording punch in/out (or other software functions) with it. Use the Set buttons to determine what keystroke is triggered by the pedal-up and pedal-down positions. You can assign the pedal to any two keystrokes you wish. (You are not restricted to punch in/out.)

Wave support for legacy (MME) software

Windows only exposes the first two channels of a multi-channel WDM audio stream to applications which use the legacy (MME) multimedia interface. The *Enable full Wave support for legacy (MME) software (less efficient)* option forces the 896HD multimedia driver to expose all channels as stereo pairs, providing full MME support.

If your host audio software does *not* directly support WDM audio and instead only supports legacy MME drivers, use this option to access multiple 896HD input and output channels.

If your host audio software *does* directly support WDM audio, leave this option unchecked for optimal performance.

This option is only available when the multimedia driver has been installed, and it defaults to being not checked.

Multichannel Wave synchronization

This option should always be left on (checked). There are only a few rare cases in which you would want to turn it off. For details, refer to the MOTU tech support database at www.motu.com.

896HD TAB SETTINGS

Optical input/output

The *Optical input* and *Optical output* settings let you choose between ADAT ('lightpipe') or OFF. Turning it off frees up FireWire bandwidth. In other words, it opens up resources on the FireWire bus for other devices connected via FireWire.

Phones

The *Phones* setting lets you choose what you will hear from the headphone jack. Choose *Main Outs* if you'd like the headphone output to match the main outs. Choose *Phones* if you would like the headphones to serve as their own independent output, which you can access as an independent output destination in your host audio software and as an output destination for the four on-board CueMix DSP mix busses.

At the 4x sample rates (176.4 and 192kHz), the headphone output can be assigned to any analog output pair or the *Phones* setting, as described above. But at the 4x sample rates, the *Phones* output is not available as an output destination for software on the computer. Instead, it is only available as a destination for the two CueMix DSP mixes. In other words, it can only take CueMix inputs.

Sample Rate Conversion

This option lets you control AES/EBU sample rate conversion. Sample rate conversion is available when the 896HD is operating at the 1x sample rates (44.1 and 48kHz) or the 2x sample rates (88.2 or 96kHz). AES/EBU is disabled entirely at the 4x sample rates (176.4 and 192kHz). Each option is explained below.

None

No sample rate conversion occurs. Both the AES/EBU input and output match the sample rate of the 896HD's system clock.

AES In

The AES/EBU input locks to the sample rate of the input signal (whatever it happens to be) and converts it to the 896HD system clock rate. The Rate Conversion LEDs on the 896HD front panel indicate the incoming sample rate and that rate conversion is occurring.

AES Out slave to AES in

To make the AES/EBU output sample rate match the sample rate currently being received by the 896HD's AES/EBU input, choose *AES Out slave to AES in*. This setting requires a connection to the 896HD's AES/EBU input from a device that is transmitting an AES/EBU clock signal.

⚠ Be careful when both the 896HD's AES/EBU input and output are connected to the same external device: this option is likely to create a clock loop.

⚠ When you are using the AES/EBU input as a clock source for sample rate conversion on the AES/EBU output, you cannot use the AES/EBU input for audio input.

AES Out x 2 / AES Out ÷ 2

Choose one of these sample rates when the desired AES/EBU output rate needs to be twice the 896HD system clock rate (when the system clock is at either 44.1 or 48 kHz) or half the system clock rate (when the system clock is at 88.2 or 96 kHz). Either way, the AES/EBU output remains resolved to the 896HD system clock. For further details about this option, see "Syncing AES/EBU devices" on page 28.

Programmable Meters

This option lets you choose which bank you wish to monitor with the eight programmable meters on the MOTU 896HD front panel. Your choices are: Analog Out, ADAT In or ADAT Out. You can also adjust this setting by repeatedly pushing the VOLUME knob on the 896HD front panel.

AES/EBU Meters

This option lets you choose to monitor either AES/EBU input or output with the programmable AES/EBU meters on the MOTU 896HD front panel. You can also adjust this setting by repeatedly pushing the VOLUME knob on the 896HD front panel.

Clip Hold Time

The *Clip Hold Time* option controls how long the top red LED remains illuminated after clipping occurs (see Figure 5-2 below).

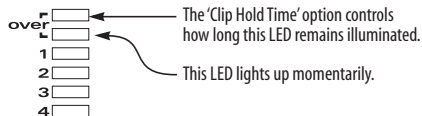


Figure 5-2: The *Clip Hold Time* option.

Peak Hold Time

The 896HD front-panel level meters support standard peak/hold metering, where the LED for the highest level recently measured on the channel remains illuminated for a brief period of time while the rest of the LEDs below it remain fully dynamic. The *Peak Hold Time* controls how long the peak-hold LED remain illuminated before going dark again.

Disable Interface option

Check this option to temporarily take an interface off line. Doing so frees up FireWire bandwidth for other devices without turning off the disabled 896HD.

Word Out

The *Word Out* menu appears when the 896HD is operating at a 2x sample rate (88.2 or 96kHz) or 4x sample rate (176.4 or 192kHz). This menu lets you set the word clock output either to match the current sample rate (*System Clock*) or force it to the corresponding 1x rate (either 44.1 or 48kHz). For example, if the 896HD were operating at 176.4kHz, choosing the *Force 44.1/48kHz* option would produce word clock output at 44.1kHz.

CHAPTER 6 Cubase, Nuendo and Other ASIO Software

OVERVIEW

The 896HD includes a ASIO driver that provides multi-channel I/O and sample-accurate synchronization with Steinberg’s Cubase family of digital audio sequencers, including Cubase VST, Cubase SX and Nuendo. The 896HD requires Cubase for Windows Version 3.553 or higher.

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WHAT IS ASIO?

ASIO is an acronym for *Audio Streaming Input and Output*. The ASIO MOTU FireWire Audio driver allows the 896HD to provide multi-channel audio input and output for any audio application that supports ASIO drivers.

Sample-accurate sync

The MOTU FireWire Audio ASIO driver supports sample-accurate sync (via the 896HD’s ADAT sync feature) for applications that support it.

Attention: Cubase VST users

Cubase VST Version 5 is used for the examples in this chapter. However, there is no significant difference between the Version 5 examples shown and what you see in Version 4. The basic procedures are the same.

Attention: Other software users

The 896HD ASIO driver also provides multi-channel I/O with any ASIO-compatible audio software. Cubase is used for the examples in this chapter. However, the basic procedures are the same and can be easily applied to any ASIO-compatible software. Just follow the general descriptions at the beginning of each main section in this chapter. Consult your software documentation for details about each topic, if necessary.

If your audio software doesn’t support ASIO

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

PREPARATION

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

- chapter 4, “Installing the 896HD Hardware” (page 17).
- chapter 3, “IMPORTANT! Run the 896HD Software Installer First” (page 15)

RUN THE MOTU FIREWIRE AUDIO CONSOLE

Before you run Cubase, launch the MOTU FireWire Audio Console to configure your 896HD hardware. The MOTU FireWire Audio Console lets you configure your audio interface, and it lets you

enable the desired inputs and outputs. Only enabled inputs and outputs will be available to Cubase, so this is an important step. For complete details regarding the MOTU FireWire Audio Console, see chapter 5, “MOTU FireWire Audio Console” (page 35).

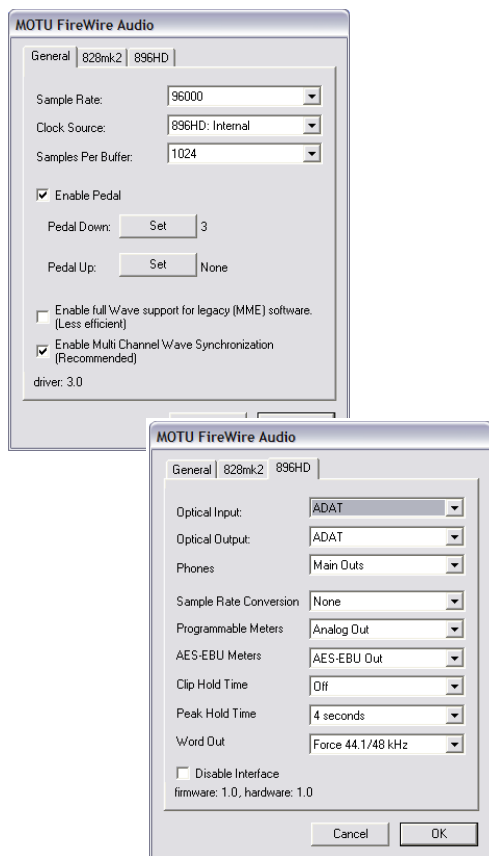


Figure 6-1: The MOTU FireWire Audio Console gives you access to all of the settings in the 896HD hardware, including the clock source, sample rate and optical I/O enable/disable.

For complete details about the 896HD settings, see chapter 6, “MOTU FireWire Control Panel (Mac OS 9)” (page 45). The following sections provide a brief explanation of each 896HD setting for use with Cubase.

Sample rate

Choose the desired overall sample rate for the 896HD system and Cubase. Newly recorded audio in Cubase will have this sample rate. Before running the 896HD at the 4x sample rates, see “Operation at 4x sample rates (176.4 or 192kHz)” on page 36.

Clock Source

This setting is very important because it determines which audio clock the 896HD will follow.

If you do not have any digital audio connections to your 896HD (you are using the analog inputs and outputs only), and you will not be slaving Cubase to an external clock source, choose *Internal*.

If you are slaving the 896HD to the ADAT Sync or Word Clock input connector, choose *ADAT 9-pin* or *Word Clock In*, respectively.

For information about the other clock source settings, see “Clock Source” on page 37.

If you have digital audio devices connected to the 896HD, see “Making sync connections” on page 21.

Samples Per Buffer

The *Samples Per Buffer* setting can be used to reduce the delay — or *monitoring latency* — that you hear when live audio is patched through your 896HD hardware and Cubase. For example, you might have MIDI instruments, samplers, microphones, and so on connected to the analog inputs of the 896HD. If so, you will often be mixing their live input with audio material recorded in Cubase. See chapter 8, “Reducing Monitoring Latency” (page 59) for complete details.

Optical input and output

To make the 896HD optical input or output available in Cubase, turn them on in the optical input and/or output menu. If you won't be using the optical connectors, turn them off.

Phones

This 896HD setting lets you choose what you'll hear from the headphone jack. For example, if you choose *Main Outs*, the headphones will duplicate the main outs. Or you can choose any other output pair. If you choose *Phones*, this setting makes the headphone jack serve as its own independent output pair (except when running at 176.4 or 192kHz). As a result, you'll see *Phones 1-2* as an additional audio destination in Cubase's audio output menus.

CHOOSING THE MOTU FIREWIRE ASIO DRIVER

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

Cubase VST

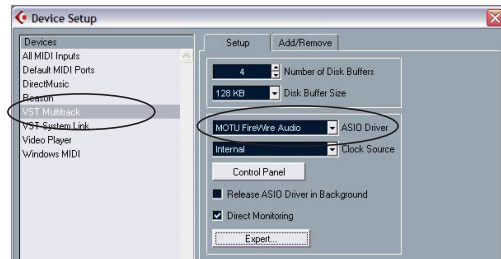
To activate the 896HD driver in Cubase VST, choose *Audio Setup>System* from the Options menu, and then choose *MOTU FireWire ASIO* from the ASIO device menu. Make the other settings in the dialog as needed for your system and synchronization scenario.

Nuendo and Cubase SX

To activate the 896HD driver in Nuendo or Cubase SX, go to the Device Setup window, click VST Multitrack and choose *MOTU FireWire ASIO* from

the ASIO Driver menu as shown below. Make the other settings in the dialog as needed for your system and synchronization scenario.

Nuendo and Cubase SX



Cubase VST

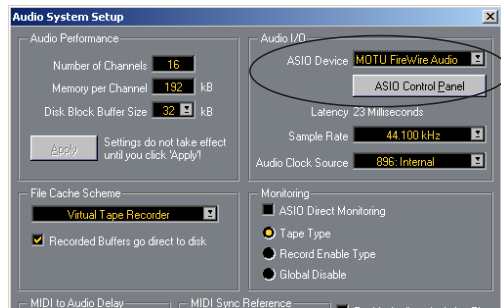


Figure 6-2: Activating the 896HD FireWire ASIO driver in Nuendo and Cubase.

NUMBER OF CHANNELS

In Cubase, be sure to choose enough channels in the System dialog (as shown above in Figure 6-2) to cover the 18 channels of input and 22 channels of output provided by your 896HD — although the number of channels may depend on how your 896HD is configured:

- 12 channels for analog I/O (including the headphone out)
- 2 channels for AES/EBU
- Zero or 8 channels for optical, depending on whether you have optical turned off or set to *ADAT optical*

For example, if you are using analog only, the 896HD requires a minimum of 12 channels (for 12 channels of output). If you are using analog and AES/EBU, you need 14 channels.

As another example, if you are using analog, AES/EBU and ADAT optical, you need 22 channels (the maximum number of simultaneous output channels provided by the 896HD).

In Cubase, set the number of channels in the System dialog (as shown above in Figure 6-2).

ASIO DIRECT MONITORING

The ASIO Direct Monitoring option (Figure 6-2) allows you to monitor inputs directly in the 896HD hardware with no drain on your computer and near zero latency. When you enable this option, Cubase uses the 896HD's CueMix DSP monitoring features whenever you use Cubase's monitoring features. For further information, see "Controlling CueMix DSP from within Cubase or Nuendo" on page 63.

OTHER SYSTEM DIALOG SETTINGS

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

ACTIVATING 896HD INPUTS

Once you've chosen the MOTU FireWire ASIO driver in the Audio System dialog as explained earlier in "Choosing the MOTU FireWire ASIO driver" on page 45, choose *VST Inputs* from the *Panels* menu (or the *Devices* menu in Cubase SX) to see the 896HD inputs. To activate them, click the *Active* light next to each input. If you don't see the optical inputs and/or outputs, check the MOTU FireWire Audio Console to make sure they are turned on and set to the format you require. If you don't plan to use the optical input or output, turn it off to conserve computer bandwidth.

The "Mix1 1-2" input

In Cubase's VST Inputs window, you'll see an 896HD input called *Mix1 1-2*. This input source delivers the output of CueMix DSP "MIX1" (the first mix bus of the four on-board no-latency monitor mixes in the 896HD) to your computer. This input serves, for example, as a convenient way for you to record the 896HD's MIX1 monitor mix into Cubase (for reference and archiving purposes). Further, if you are sending audio from Cubase to the same output pair as MIX1, you can choose to either include or exclude the audio from the computer in the Mix1 stream being sent to Cubase. For details on how to do this, see "Mix1 Return Includes Computer" on page 68.

The Mix1 1-2 input is not available at the 4x sample rates (176.4 or 192kHz).

⚠ Warning: the Mix1 1-2 input can cause feedback loops! DO NOT assign this input to a track that shares the same 896HD output pair as MIX1.



Figure 6-3: Activating 896HD inputs in Cubase VST.



Figure 6-4: Activating 896HD inputs in Nuendo or Cubase SX.

ASSIGNING INPUTS

Once you've activated the 896HD inputs as shown in the previous section, you can then assign them to Cubase or Nuendo audio channels in the channel mixers in the usual fashion.

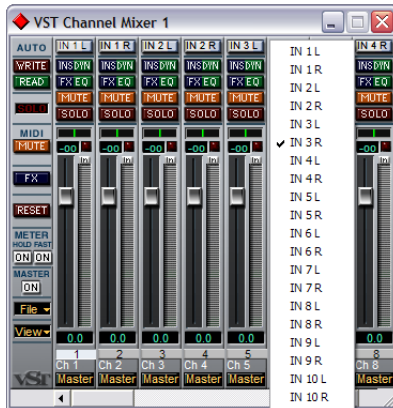


Figure 6-5: To assign an 896HD input to a Cubase VST audio channel: control-click the input button at the top of the channel strip. For Nuendo or Cubase SX, consult your documentation.

TRIMMING THE ANALOG INPUTS

The 896HD analog inputs provide trim knobs on the front panel. To calibrate an audio input, feed signal to the input, and adjust the input's corresponding trim knob on the front panel of the 896HD until peaks in the level meter are as high as possible without clipping (hitting zero dB).

ASSIGNING OUTPUTS

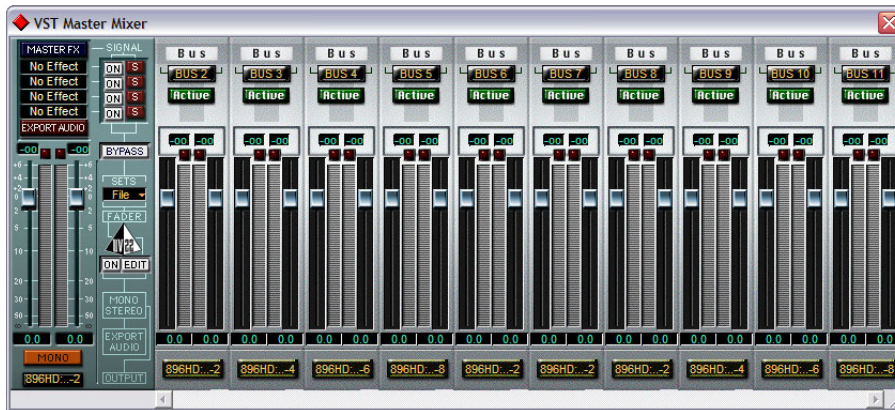
Once you've chosen the MOTU FireWire ASIO driver in the Audio System dialog as explained earlier in "Choosing the MOTU FireWire ASIO driver" on page 45, 896HD outputs will be available in Cubase or Nuendo as output destinations. In Cubase VST, these outputs appear in the VST Master Mixer window as output

assignments for the master fader and busses, as shown below in Figure 6-6. In NuendoCubase SX, they appear in the VST Outputs window.

The "Phones 1-2" output

If you've chosen to treat the 896HD headphones as an independent output, you'll see *Phones 1-2* as an 896HD output destination. Audio tracks assigned

In Cubase VST, use the output buttons at the bottom of each channel strip, including the master fader, to assign 896HD outputs to busses. You can then assign channels in the VST Master Mixer window to each bus as desired.



In Nuendo or Cubase SX, access the 896HD outputs via the busses in the VST Outputs window.

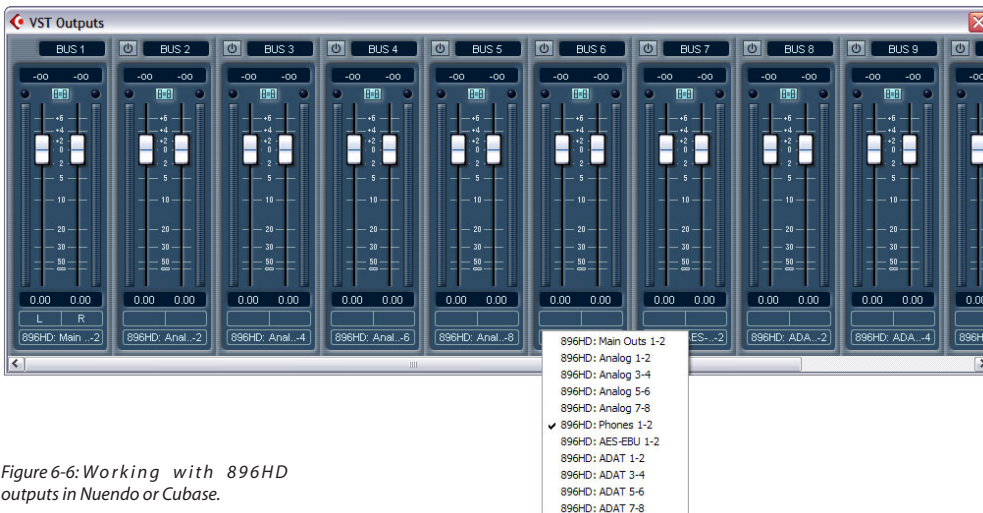


Figure 6-6: Working with 896HD outputs in Nuendo or Cubase.

to this output pair will be heard on the headphone jack only. For further explanation, see “Phones” on page 45.

CHANGING 896HD SETTINGS

To change the 896HD settings at any time, go to the Audio System Setup window in Cubase VST or the Device Setup window in Nuendo or Cubase SX and click the ASIO Control Panel button, as shown in Figure 6-2 on page 45.

PROCESSING LIVE INPUTS WITH PLUG-INS

If you patch a live input (such as a MIDI synthesizer) through a VST plug-in effect in Cubase, you might hear a slight delay. There are several ways to reduce this delay. For details, see chapter 8, “Reducing Monitoring Latency” (page 59).

SYNCHRONIZATION

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

As you read through the following sections to decide what form of synchronization you might need with other devices in your studio, be sure to consult chapter 4, “Installing the 896HD Hardware” (page 17) for the proper hardware connections. Use the synchronization diagrams in that chapter to be clear about how you will be synchronizing Cubase to the other components of your system.

Running Cubase or Nuendo under its own transport control

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

However, even though Cubase or Nuendo is not slaving to SMPTE time code, you still need to be concerned with the synchronization of the 896HD’s digital audio clock with other devices connected to it digitally (if any). For example, if you have a digital mixer connected to an 896HD interface via an ADAT optical lightpipe cable, you need to make sure that their audio clocks are phase-locked. For details, see “Syncing optical devices” on page 27 and “Making sync connections” on page 21. If you don’t have any digital audio devices connected, digital audio phase-lock does not apply to you.

Resolving Cubase or Nuendo and the 896HD to word clock, video and/or SMPTE time code

To resolve your 896HD to word clock, video and/or SMPTE time code using an additional synchronization device, use the setup shown in “Syncing word clock devices” on page 31 or “Syncing to video and/or SMPTE time code” on page 26.

Follow the instructions in your Cubase or Nuendo manual for slaving them to MIDI Time Code (MTC). To ensure that your audio tracks don’t drift out of sync with your MIDI tracks — or time code, use a hardware synchronizer like the MIDI Timepiece AV or Digital Timepiece to resolve the 896HD hardware as well, as explained in “Syncing to video and/or SMPTE time code” on page 26. A digital audio synchronizer is required for drift-free SMPTE/MIDI time code sync. Make sure the *Clock Source* setting in the MOTU FireWire Audio Console window has the appropriate setting for locking the 896HD to the synchronizer. For example, in Figure 4-11 on page 26, word clock is being used to resolve an 896HD interface, so the *Clock Source* setting is *Word Clock In*.

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

Sample-accurate sync to ADAT or Tascam

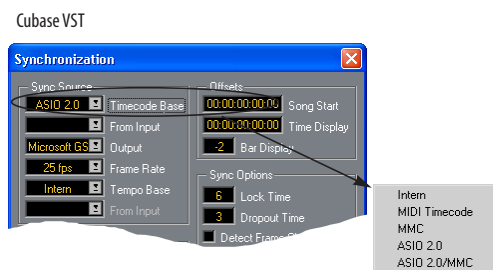
Cubase and Nuendo, along with the 896HD and its ASIO 2 driver, provide you with sample-accurate transfers with ADATs, Alesis recorders and any other devices that support standard ADAT sample address (*ADAT Sync*).

Similarly, with the help of a MOTU Digital Timepiece universal A/V synchronizer, Cubase (or Nuendo) and an 896HD can perform sample-accurate transfers with Tascam digital recorders.

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

For details on how to connect your hardware for sample-accurate sync, see “Sample-accurate sync” on page 23. Then, set up Cubase as follows:

- 1 Choose *ADAT 9-pin* as the *Audio Clock Source* setting. In Cubase VST, this setting is in the Audio System Setup window (Audio menu). In Nuendo or Cubase SX, this setting is in the Device Setup window (Options menu).
- 2 Go to Cubase or Nuendo’s *Synchronization* window, as shown below:



Nuendo or Cubase SX

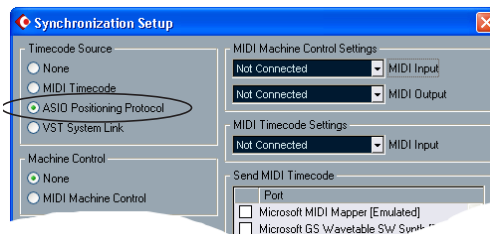


Figure 6-7: Setting up sample-accurate sync via ASIO 2.

3 If you are *not* using an MMC-compatible synchronizer (such as a MOTU MIDI Timepiece AV, Digital Timepiece or Alesis BRC), choose the settings shown above in Figure 6-7 that applies to you. In this scenario, transport control is handled by the ADAT (or other sample-accurate sync source).

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

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



Figure 6-8: Enabling the SYNC button.

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

MIDI MACHINE CONTROL (MMC)

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

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

See “Sample-accurate sync to ADAT or Tascam” on page 50 for details on how to set this up.

USING A FOOT SWITCH

Use a foot switch connected to the 896HD to trigger recording punch-in and punch-out, or any other feature in your host audio software that is assigned to a computer keystroke. To trigger a different set of keystrokes with the foot switch, visit the MOTU FireWire Audio Console. (See “Enable Pedal” on page 39.)

24-BIT OPERATION

Your 896HD hardware fully supports Cubase and Nuendo's 24-bit recording capabilities. Simply enable 24-bit operation as instructed in your Cubase or Nuendo manual. The 896HD always supplies a 24-bit data stream, and when you enable 24-bit operation in Cubase or Nuendo, it simply uses all 24-bits supplied by the 896HD hardware.

MONITORING SYSTEM PERFORMANCE

Because it has so many inputs and outputs, the 896HD may push the limits of your computer's processing power. Keep the VST Performance window open to keep tabs on the load on your CPU and disk buffers. If the meters get too high, you can reduce the load by reducing the number of inputs and outputs you are working with. Use the MOTU FireWire Audio Console to uncheck input check boxes and set output source menus to *None*.

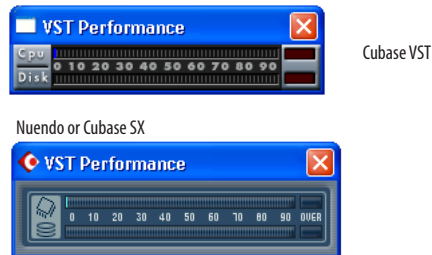


Figure 6-9: Keep the Audio Performance window open to keep tabs on your computer's processing power and hard disk performance.

CHAPTER 7 **Sonar and other WDM Software**

OVERVIEW

The 896HD WDM driver provides standard multi-channel input and output for WDM-compatible audio software running under Windows Me, 2000 and XP.

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WDM AND WAVE DRIVER COMPATIBILITY

WDM is an acronym for *Windows Driver Model*. The MOTU FireWire WDM driver allows the 896HD to provide multi-channel audio input and output for any audio application that supports WDM audio drivers.

Attention: Cakewalk users

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

Attention: other software users

Cakewalk's SONAR software is used for the WDM driver setup examples in this chapter. However, the basic procedures are the same and can be easily applied to any WDM-compatible software. Just follow the general descriptions at the beginning of each main section in this chapter. Consult your software documentation for details about each topic, if necessary.

INSTALLING THE MOTU FIREWIRE AUDIO WDM DRIVER

The MOTU FireWire Audio Installer CD installs the MOTU FireWire WDM driver into Windows for you. That's pretty much all the preparation you need. See chapter 3, "IMPORTANT! Run the 896HD Software Installer First" (page 15).

ENABLING WAVE (MME) COMPATIBILITY

If your host audio software does not yet support Windows WDM drivers, enable Wave driver compatibility in the MOTU FireWire WDM driver so that all of the 896HD's inputs and outputs show up in your software. For details, see "Wave support for legacy (MME) software" on page 39.

☛ If your host audio software *does* support WDM audio drivers, *don't* enable Wave driver compatibility.

MAKING SETTINGS IN THE MOTU FIREWIRE AUDIO CONSOLE

Before you run your audio software, launch the MOTU FireWire Audio Console to configure your 896HD hardware. The MOTU FireWire Audio Console lets you choose the audio clock source, and it lets you enable the optical input and output, if needed. For details, see chapter 5, “MOTU FireWire Audio Console” (page 35).

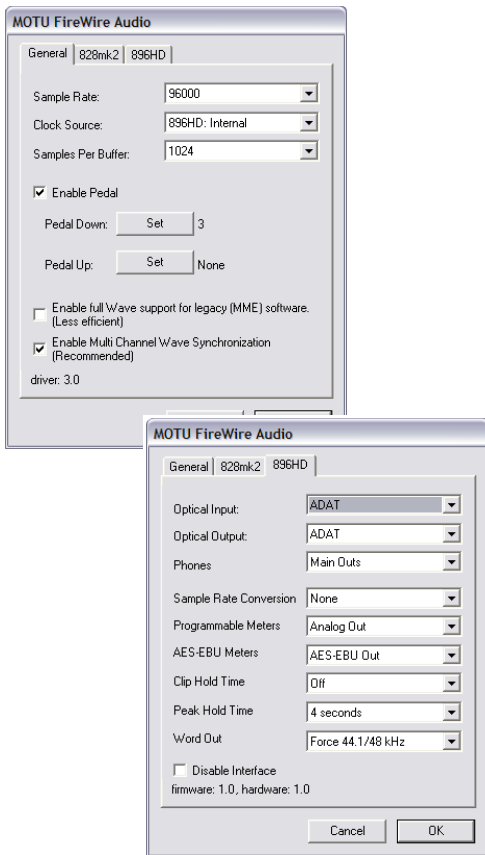


Figure 7-1: The MOTU FireWire Audio Console gives you access to all of the settings in the 896HD hardware, including the clock source, sample rate and optical I/O enable/disable.

For complete details about the 896HD settings, see chapter 5, “MOTU FireWire Audio Console” (page 35). The following sections provide a brief explanation of each 896HD setting for use with Sonar.

Sample rate

Choose the desired overall sample rate for the 896HD system and Sonar. Newly recorded audio in Sonar will have this sample rate.

Before running the 896HD at the 4x sample rates, see “Operation at 4x sample rates (176.4 or 192kHz)” on page 36.

Clock Source

This setting is very important because it determines which audio clock the 896HD will follow.

If you do not have any digital audio connections to your 896HD (you are using the analog inputs and outputs only), and you will not be slaving Sonar to external SMPTE time code, choose *Internal*.

If you are slaving the 896HD to the ADAT Sync or Word Clock input connector, choose *ADAT 9-pin* or *Word Clock In*, respectively.

For information about the other clock source settings, see “Clock Source” on page 37.

If you have digital audio devices connected to the 896HD, see “Making sync connections” on page 21.

Samples Per Buffer

The *Samples Per Buffer* setting can be used to reduce the delay — or *monitoring latency* — that you hear when live audio is patched through your 896HD hardware and Sonar. For example, you might have MIDI instruments, samplers, microphones, and so on connected to the analog inputs of the 896HD. If so, you will often be mixing

their live input with audio material recorded in Sonar. See chapter 8, “Reducing Monitoring Latency” (page 59) for complete details.

Optical input and output

To make the 896HD optical input or output available in Digital Performer, turn them on in the optical input and/or output menu. If you won’t be using the optical connectors, turn them off.

Phones

This 896HD setting lets you choose what you’ll hear from the headphone jack. For example, if you choose *Main Outs*, the headphones will duplicate the main outs. Or you can choose any other output pair. If you choose *Phones*, this setting makes the headphone jack serve as its own independent output pair (except when running at 176.4 or 192kHz). As a result, you’ll see *Phones 1-2* as an additional audio destination in Sonar’s audio output menus.

ENABLING THE MOTU FIREWIRE AUDIO WDM DRIVER

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

Enabling the MOTU FireWire Audio WDM driver in SONAR

To activate the MOTU FireWire Audio WDM driver in SONAR:

- 1 Choose *Audio* from the Options menu.
- 2 Click the *Drivers* tab.
- 3 Highlight the 896HD inputs and outputs that you wish to use and unhighlight the ones you don’t as shown in Figure 7-2.

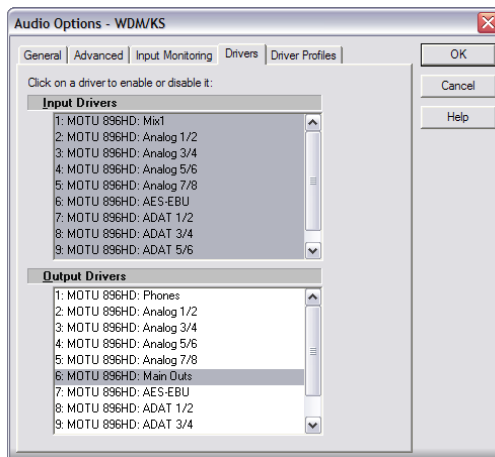


Figure 7-2: Activating the MOTU FireWire Audio WDM driver in SONAR by highlighting 896HD inputs and outputs.

Other audio settings in SONAR

There are a few other settings in SONAR that impact 896HD operation:

- 1 In the Audio Options window, click the *General* tab.
- 2 Choose a 896HD input and output for recording and playback timing as shown below in Figure 7-3. It doesn’t matter which input or output you choose.

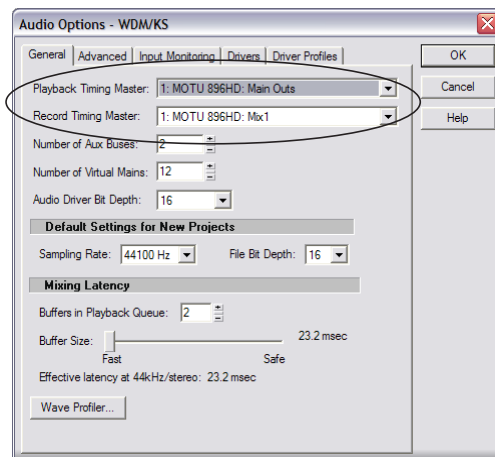


Figure 7-3: Make sure you have chosen a 896HD input and output for the playback and recording timing master settings.

3 In the General tab, click the Wave Profiler button and run the Wave Profiler.

This process chooses the optimum settings for the 896HD hardware.

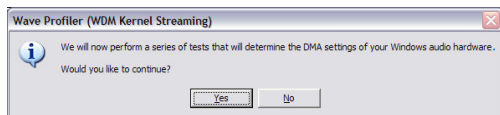


Figure 7-4: Running the Wave Profiler in Cakewalk's SONAR.

NUMBER OF CHANNELS

In the Audio Options window (Figure 7-3), be sure to choose enough *Virtual Mains* in the *General* tab to cover the 18 channels of input and 22 channels of output provided by your 896HD — although the number of channels may depend on how your 896HD is configured:

- 12 channels for analog I/O (including the headphone out)
- 2 channels for AES/EBU
- Zero or 8 channels for optical, depending on whether you have optical turned off or set to *ADAT optical*

For example, if you are using analog only, the 896HD requires a minimum of 12 voices (for 12 channels of output). If you are using analog and AES/EBU, you need 14 voices.

As another example, if you are using analog, AES/EBU and ADAT optical, you need 22 voices (the maximum number of simultaneous output channels provided by the 896HD).

OTHER AUDIO OPTIONS

Consult your sonar documentation for details about the rest of the settings in this dialog.

WORKING WITH 896HD INPUTS AND OUTPUTS

Once you've enabled the 896HD inputs and outputs in the Drivers tab of the Audio Options window (Figure 7-2 on page 55), 896HD audio inputs and outputs will appear in Sonar's input/output menus, and you can set them up and use them as any standard audio input and output. If you don't see the optical inputs and/or outputs, check the MOTU FireWire Audio Console to make sure they are turned on and set to the format you need. If you don't plan to use the optical input or output, turn it off to conserve computer bandwidth.

Trimming the analog inputs

The 896HD analog inputs provide trim knobs on the front panel. To calibrate an audio input, feed signal to the input, and adjust the input's corresponding trim knob on the front panel of the 896HD until peaks in the level meter are as high as possible without clipping (hitting zero dB).

The "Mix1 1-2" input

In Sonar's Input Drivers list, you'll see an 896HD input called *Mix1 1-2*. This input source delivers the output of CueMix DSP "Mix 1" (the first mix bus of the four on-board no-latency monitor mixes in the 896HD) to your computer. This input serves, for example, as a convenient way for you to record the 896HD's Mix 1 monitor mix into Sonar (for reference and archiving purposes). Further, if you are sending audio from Sonar to the same output pair as MIX1, you can choose to either include or exclude the audio from the computer in the MIX1 stream being sent to Sonar. For details on how to do this, see "Mix1 Return Includes Computer" on page 68.

⚠ Warning: the Mix1 1-2 input can cause feedback loops! DO NOT assign this input to a track that shares the same 896HD output pair as MIX1.

CHANGING 896HD SETTINGS

You can change the 896HD settings at any time by accessing the MOTU FireWire Audio Console.

REDUCING DELAY WHEN MONITORING LIVE INPUTS

If you have live audio inputs connected to the 896HD, such as MIDI synthesizers, samplers, microphones or other live instruments, you might hear a slight delay when their audio is being monitored through your 896HD hardware and your host audio program. There are several ways to reduce — and eliminate — this audible monitoring delay. For details, see chapter 8, “Reducing Monitoring Latency” (page 59).

SYNCHRONIZATION

If your audio software has the ability to slave to SMPTE time code, then it can take advantage of the synchronization features in the 896HD.

As you read through the following sections to decide what form of synchronization you might need with other devices in your studio, be sure to consult “Making sync connections” on page 21 for the proper hardware connections. Use the synchronization diagrams to be clear about how you will be synchronizing your audio software and the 896HD to the other components of your system.

Synchronizing digital audio connections

If you have devices connected to the 896HD digital inputs (optical or AES/EBU), you need to be concerned with the synchronization of the 896HD’s digital audio clock with other devices connected to it digitally (if any). For example, if you have a digital mixer connected to the 896HD via an ADAT optical light pipe cable, you need to make sure that their audio clocks are phase-locked. For details, see “Syncing optical devices” on

page 27 and “Making sync connections” on page 21. If you don’t have any digital audio devices connected to the 896HD, digital audio phase-lock does not apply to you.

Slaving to SMPTE time code

If you need to slave your audio software and the 896HD system to SMPTE time code, follow the instructions in your software’s manual for slaving it to MIDI Time Code (MTC). To ensure that your audio tracks don’t drift out of sync with your MIDI tracks or the time code, use a hardware synchronizer like the MOTU MIDI Timepiece AV or Digital Timepiece to slave the 896HD hardware to the SMPTE (or MIDI Time) Code as well. A digital audio synchronizer is required for drift-free SMPTE/MIDI time code sync. Make sure the *Clock Source* setting in the MOTU FireWire Audio Console is set to *Word Clock In*. For examples of how to set this up, see “Syncing to video and/or SMPTE time code” on page 26.

USING A FOOT SWITCH

Use a foot switch connected to the 896HD to trigger recording punch-in and punch-out, or any other feature in your host audio software that is assigned to a computer keystroke. By default, the foot switch triggers the 3 key on the computer keypad. To trigger a different set of keystrokes with the foot switch, visit the MOTU FireWire Audio Console. (See “Enable Pedal” on page 39.)

24-BIT OPERATION

Your 896HD hardware fully supports 24-bit recording in any audio software that supports it. Simply enable 24-bit operation as instructed by the software. The 896HD system always supplies the software with a 24-bit data stream, and when you enable 24-bit operation, it simply uses all 24-bits supplied by the 896HD hardware.

CHAPTER 8 Reducing Monitoring Latency

OVERVIEW

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

This delay is caused by the amount of time it takes for audio to make the entire round trip through your computer, from when it first enters an 896HD input, passes through the 896HD hardware into the computer, through your host audio software, and then back out to an 896HD output.

If you don't need to process a live input with plug-ins, the easiest way to avoid monitoring latency is to use the 896HD's CueMix DSP feature to patch the input directly to your monitor outs via the 896HD audio hardware. This is just like bussing inputs to outputs in a digital mixer. For details, see "CueMix DSP hardware monitoring" on page 62.

If you *do* need to process a live input with plug-ins, or if you are playing virtual instruments live through your 896HD audio hardware, you can significantly reduce latency — and even make it completely inaudible, regardless of what host audio application software you use. This chapter explains how.

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

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MONITORING LIVE INPUT

There are two ways to monitor live audio input with an 896HD: 1) through the computer or 2) via CueMix™ DSP hardware monitoring. Figure 8-1 on page 60 shows method 1, which allows you to add effects processing such as reverb and guitar amp effects via plug-ins in your audio software. See the next section, “Adjusting the audio I/O buffer” for details about how to reduce — and possibly eliminate — the audible monitoring delay that the computer introduces.

Figure 8-2 shows how to use CueMix™ DSP hardware-based monitoring, which lets you hear what you are recording with no monitoring delay and no computer-based effects processing. (You can add effects later, after you’ve recorded the live input as a disk track.) See “CueMix DSP hardware monitoring” later in this chapter for details on how to use CueMix DSP with your audio software, or with the included CueMix Console software.

If the material you are recording is suitable, there is a third way to monitor live input: use both methods (Figure 8-1 and Figure 8-2) at the same time. For example, you could route vocals to both the

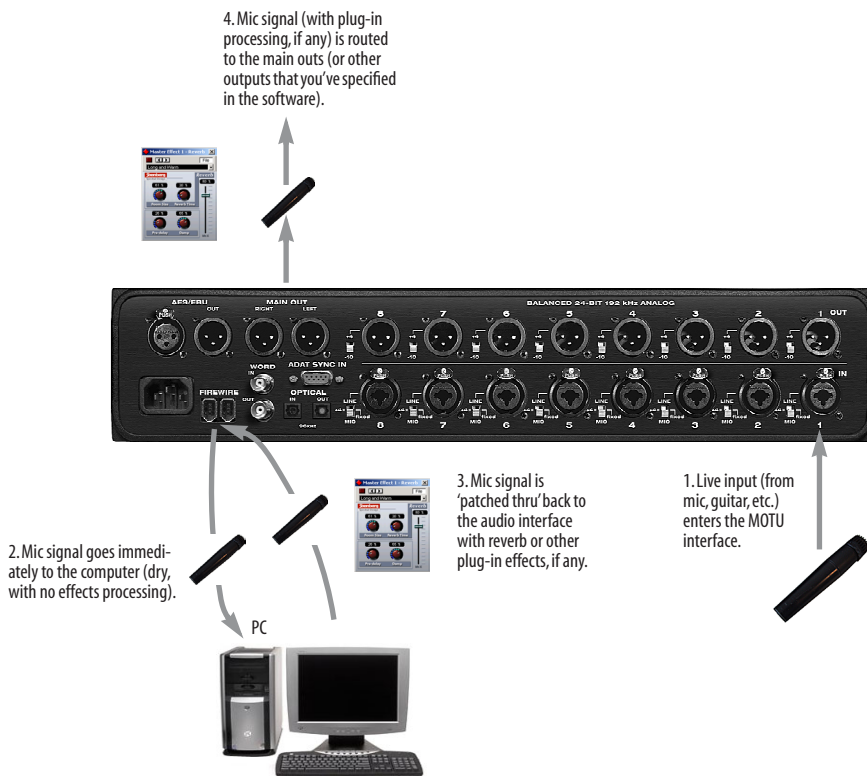
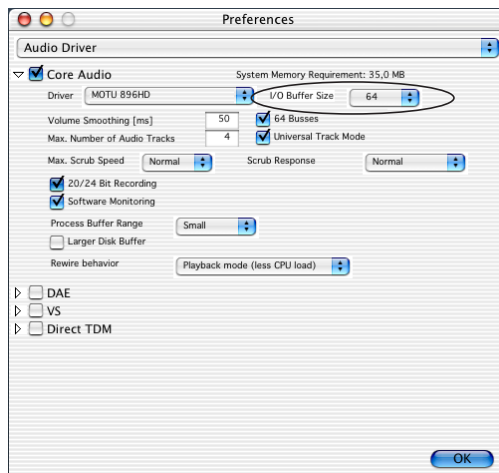


Figure 8-1: There are two ways to monitor live audio inputs with an 896HD: 1) through the computer or 2) via CueMix™ DSP hardware monitoring. This diagram shows method 1 (through the computer). When using this method, use your host software's buffer setting to reduce the slight delay you hear when monitoring the live input, but don't lower it too much, or your computer might get sluggish.

computer (for a bit of reverb) and mix that processed signal on the main outs with dry vocals from CueMix DSP.

ADJUSTING THE AUDIO I/O BUFFER

A *buffer* is a small amount of computer memory used to hold data. For audio interfaces like the 896HD, buffers are used for the process of transferring audio data in and out of the computer. The size of the buffers determines how much delay you hear when monitoring live inputs through your audio software: larger buffers produce more delay; smaller buffers produce less.



Buffer size adjustment is made in the MOTU FireWire Audio Console, as shown in Figure 8-3 via the *Samples Per Buffer* setting.

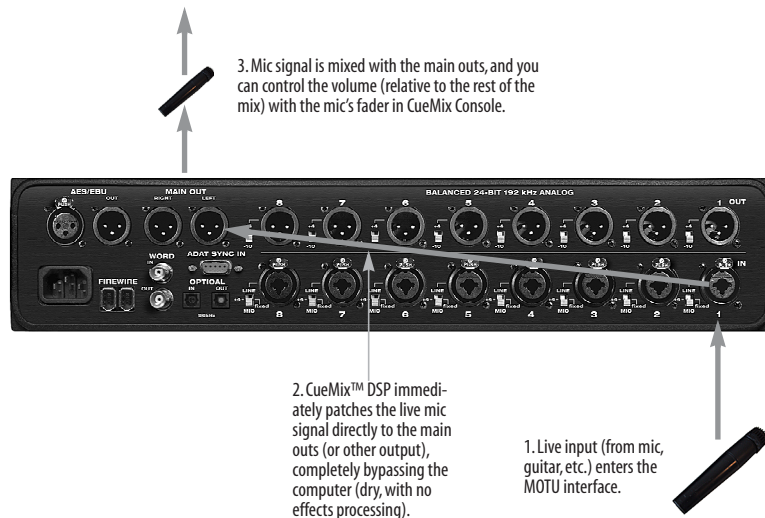


Figure 8-2: This diagram shows the signal flow when using CueMix™ DSP no-latency monitoring. Notice that this method does not allow you to process the live input with plug-ins in your audio software while it is being monitored. You can, however, add effects later — after recording the live input as a disk track. CueMix™ DSP lets you hear what you are recording with no delay and no computer-based effects.

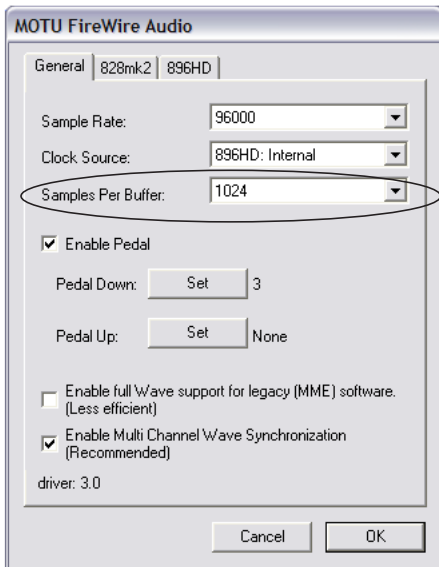


Figure 8-3: Lowering the 'Samples Per Buffer' setting in the MOTU FireWire Audio Console Window reduces patch thru latency. But doing so increases the processing load on your computer, so keep an eye on the Performance Monitor window in AudioDesk your host audio software.

Lower latency versus higher CPU overhead

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

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

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

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

Transport responsiveness

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

Effects processing and automated mixing

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

CUEMIX DSP HARDWARE MONITORING

The 896HD has a more direct method of patching audio through the system. This method is called *CueMix DSP*. When enabled, CueMix activates hardware patch-thru in the 896HD itself. CueMix DSP has two important benefits:

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

The trade-off, however, is that CueMix DSP bypasses your host audio software. Instead, live audio inputs are patched directly through to outputs in the 896HD itself and are mixed with disk tracks playing back from your audio software. This means that you cannot apply plug-ins, mix automation, or other real-time effects that your

audio software provides. But for inputs that don't need these types of features, CueMix DSP is the way to go.

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

TWO METHODS FOR CONTROLLING CUEMIX DSP

There are two ways to control CueMix DSP:

- With CueMix Console
- From within your host audio software (if it supports direct hardware monitoring)

You can even use both methods simultaneously.

Using CueMix Console

If your host audio software does not support direct hardware monitoring, you run CueMix Console side-by-side with your audio software and manage your monitor mix in CueMix Console.

CueMix Console allows you to create up to four separate 896HD monitor mixes, or any other desired routing configurations. These routings are independent of your host audio software. For complete details, see chapter 9, “CueMix Console” (page 65).

Controlling CueMix DSP from your audio software

Some ASIO-compatible audio applications, such as Cubase and Nuendo, allow you to control CueMix DSP monitoring from within the application (without the need to use CueMix Console). In most cases, this support consists of patching an 896HD input directly to an output when you record-arm a track. Exactly how this is handled depends on the application.

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

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

Controlling CueMix DSP from within Cubase or Nuendo

To turn on CueMix in Cubase VST, enable the *ASIO Direct Monitor* check box in the Monitoring section of the Audio System Setup window (Figure 6-2 on page 45). In Cubase SX or Nuendo, enable the Direct Monitoring check box in the Device Setup VST Multitrack tab (Figure 6-2 on page 45).

Other ASIO 2.0-compatible host software

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

Using CueMix DSP with WDM- or Wave-compatible software

Run CueMix Console (chapter 9, “CueMix Console” (page 65)) and use it to route live inputs directly to outputs, control their volume and panning, etc. If necessary, you can save your CueMix Console setup with your project file.

CHAPTER 9 CueMix Console

OVERVIEW

CueMix Console provides access to the flexible on-board mixing features of the 896HD. CueMix lets you route any combination of inputs to any stereo output pair. These mixes can be set up entirely independently of your host audio software. CueMix allows you to set up four completely independent mix configurations with the 896HD. You can also save and load mix configurations.

CueMix Console can be used independently of host audio software, or together with it. CueMix mixing dovetails with the direct monitoring (hardware patch thru) features of your host audio software, allowing you to seamlessly mix in both environments. CueMix Console can also be controlled from automated hardware control surfaces such as the Mackie Control™.

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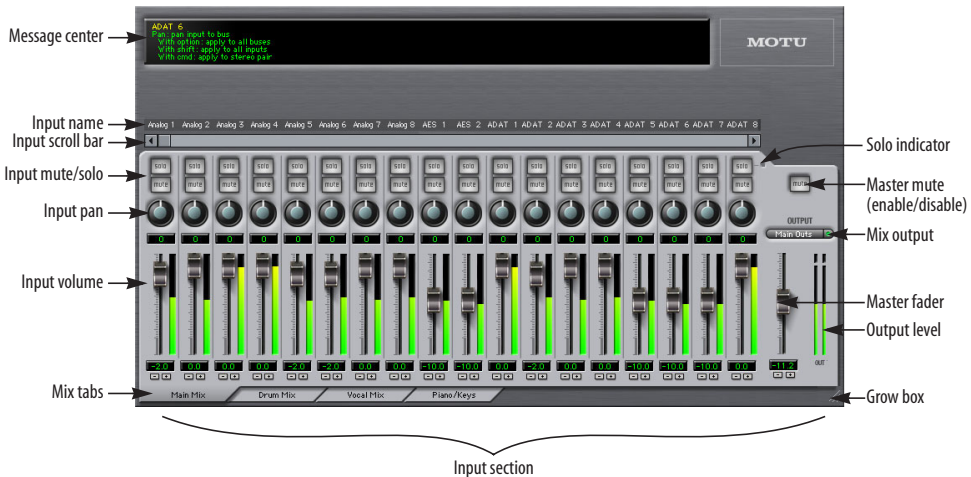


Figure 9-1: CueMix Console is a virtual mixer that gives you control over the 896HD's on-board mixing features.

ADVANTAGES OF CUEMIX MONITORING

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

- CueMix has no buffer latency. Thanks to the 896HD's DSP chip, CueMix provides the same throughput performance as a digital mixer.
- CueMix imposes absolutely no processor drain on the computer's CPU.
- CueMix routing can be maintained independently of individual software applications or projects.
- CueMix routing can operate without the computer, allowing the 896HD to operate as a portable, stand-alone mixer.

CueMix Console does not provide effects processing. For information about using your audio software's native plug-ins together with CueMix, see chapter 8, "Reducing Monitoring Latency" (page 59).

CUEMIX CONSOLE INSTALLATION

CueMix Console is installed with the rest of your 896HD software.

CUEMIX CONSOLE BASIC OPERATION

The CueMix console is simple to operate, once you understand these basic concepts.

Four mixes

CueMix provides four separate mixes: Mix1, Mix2, Mix3 and Mix4. Each mix can have any number of inputs mixed down to any 896HD output pair that you choose. For example, Mix1 could go to the headphones, Mix2 could go to the main outs, Mix3 could go to a piece of outboard gear connected to analog outputs 7-8, etc.

At the 4x samples rates (176.4 and 192kHz), CueMix DSP supports only two independent monitor mixes, due to the extremely high bandwidth demands of these sample rates.

Many inputs to one output pair

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

Viewing one mix at a time

CueMix Console displays one mix at a time. To select which mix you are viewing, click its tab at the bottom of the window, as shown in Figure 9-1. The mix name appears in the tab. Double-click the name to change it.

Each mix is completely independent

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

Widening the CueMix Console window

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

WORKING WITH A MIX

Each mix has the following components:

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

These elements are visually grouped together in the lightly shaded area in the lower half of the CueMix Console window.

Viewing a mix

To view a mix, click its tab at the bottom of the window, as shown in Figure 9-1. The mix name appears in the tab.

Naming a mix

Double-click the mix name in the tab.

Master mute

The master mute button (Figure 9-1) temporarily disables (silences) the mix.

Master fader

The master fader (Figure 9-1) controls the overall level of the mix (its volume on its stereo output). Use the individual input faders to the left to control individual input levels.

Adjusting master faders from the front panel

You can adjust the master fader level for each mix using the MONITOR LEVEL knob on the 896HD front panel. When you turn it, the analog input meters (channels 1-4) temporarily display the current volume setting for each mix. A few seconds after you stop turning (or pushing) the knob, the meters revert back to displaying analog input levels.

The MONITOR LEVEL knob controls each CueMix master fader as follows:

Action	Result
Push the knob	To view the current fader levels for all four CueMix busses.
Turn the knob	To adjust the currently selected fader. (This is indicated by a flashing red “over” LED at the top of the meter.)
Push the knob again	To cycle to the next mix bus. The flashing red “over” LED shows you which bus you are adjusting.

When using the MONITOR LEVEL knob to view and modify the CueMix master fader levels, the 8-channel bank of programmable meters (to the right of the analog input meters) displays the output level for each mix bus as follows:

Programmable meter channels	Mix bus
1-2	Mix1
3-4	Mix2
5-6	Mix3
7-8	Mix4

Output level meters

The OUT level meters show you the output for the mix’s physical output, which may include audio from your host audio software. The clip indicators clear themselves after a few seconds.

Input section

The channel strips to the left of the master fader represent each input in your 896HD. Use the input scroll bar to view additional inputs.

Input mute/solo

To add an input to a mix, or remove it, click its MUTE button. To solo it, use its SOLO button. To toggle these buttons for a stereo pair, hold down the control key while clicking either channel. The Solo indicator LED (Figure 9-1) lights up when any input is soloed (including inputs that may currently be scrolled off-screen).

Input volume and pan

Use the input fader and pan knob (Figure 9-1) to adjust these settings for the input in the mix. Again, all settings within the gray-shaded channel strip area belong to the mix currently being viewed. Note that an input can have different settings in different mixes.

To adjust the volume or panning for a stereo input pair, hold down the control key while dragging the fader or knob for either the left or right input.

SHORTCUTS

Hold down the following modifier keys as shortcuts:

Shortcut	Result
Shift key	Applies your action to all inputs in the mix.
Control key	Applies your action to the stereo input pair
space bar	Applies your action to all busses
Double-click	Returns the control to its default value (pan center, unity gain, etc.)

COPYING & PASTING (DUPLICATING) ENTIRE MIXES

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

- 1 Select the source mix (Figure 9-1) and choose Copy from the file menu (or press control-C).
- 2 Choose the destination mix and choose Paste from the file menu (or press control-V).

MESSAGE CENTER

The Message Center displays fly-over help for items in the CueMix Console window. It also displays messages regarding the overall operation of the 896HD.

FILE MENU

The CueMix Console File menu has the following items.

Save Preset / Load Preset

The 896HD can store up to 16 presets in its on-board memory. A preset includes of all CueMix DSP settings for all for mix busses, but it excludes global settings like clock source and sample rate.

The *Load Preset* and *Save Preset* commands in the CueMix Console file menu let you name, save and load presets in the 896HD hardware.

Save / Load

The *Save* and *Load* commands in the CueMix Console File menu allow you to save 896HD presets to and from your hard drive. This allows you to save an unlimited number of 896HD presets on disk. (Use the *Load Preset* and *Save Preset* commands to get presets from — and save them to — the 896HD itself.) Choose *Save* to save the current configuration; choose *Load* to open an existing configuration that you have previously saved to disk.

Mix1 Return Includes Computer

The *Mix1 return includes computer* item in the CueMix Console File menu refers to the *Mix1* bus that the 896HD driver provides as an input to host audio software. This input source delivers the output of CueMix DSP “MIX1” (the first mix bus of the four on-board no-latency monitor mixes in the 896HD) back to your computer. This input serves, for example, as a convenient way for you to record the 896HD’s MIX1 monitor mix back into your host audio software (for reference and archiving purposes).

When the *Mix1 return includes computer* menu item is checked, any audio being sent from your audio software on the computer to the same output as Mix1 will be included in the Mix1 return bus. When it is unchecked, computer output is excluded.

This menu item is essentially a pre/post switch for the computer audio insert to the stream of audio going to Mix1’s 896HD output pair (and also back to the computer).

EDIT MENU

The commands in the Edit menu let you copy and paste entire mix bus settings. See “Copying & pasting (duplicating) entire mixes” on page 68.

PHONES MENU

The Phones menu allows you to choose what you will hear on the headphone output, just like the Phones setting the MOTU FireWire Audio Console. However, this menu provides one extra option that is exclusive to CueMix Console: *Follow Active Mix*. This menu item, when checked, causes the headphone output to mirror the output of the current mix being viewed in CueMix Console. For example, if you are currently viewing Mix3 (the Mix3 tab is active), the headphones will mirror the Mix3 output (whatever it is assigned to).

CONTROL SURFACES MENU

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

Application follows control surface

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

Share surfaces with other applications

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

make changes to CueMix Console from the control surface, just bring CueMix Console to the front (make it the active application).

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

Mackie Control Surfaces

CueMix Console includes support for the following control surface products:

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

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

Enabled

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

Configure...

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

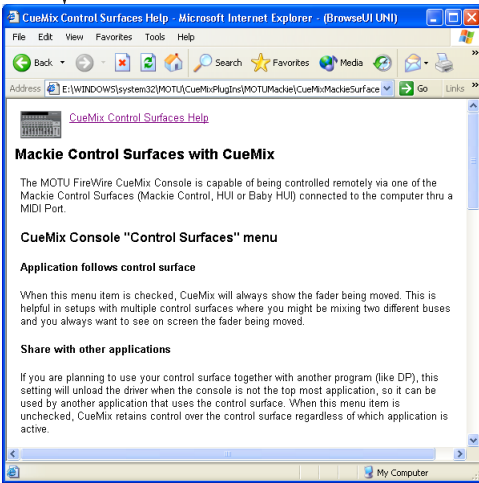
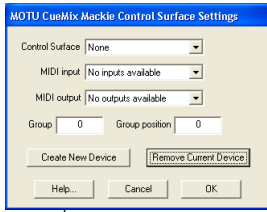


Figure 9-2: Refer to the extensive on-line help for details about configuring CueMix Console for operation with your control surface product.

Radikal Technologies SAC-2.2

The Radikal Technologies SAC-2.2 can be used via its HUI emulation mode. Just put the SAC-2.2 hardware into HUI emulation mode, and then follow the CueMix Console on-line help instructions (Figure 9-2) for HUI operation. Consult the SAC-2.2 manual for details about how put it into HUI emulation mode.

Other HUI-compatible control surfaces

Any control surface that has the ability to emulate a HUI should be compatible with CueMix Console. Just put the control surface hardware into HUI emulation mode, and then follow the CueMix Console on-line help instructions (Figure 9-2) for HUI operation. Consult the manual for the control surface for details about how put it into HUI emulation mode.

Other control surface hardware products

If you install other control surface drivers written for CueMix Console, they will appear as separate menu items at the bottom of the Control Surfaces menu, with the same sub-menu items described above.

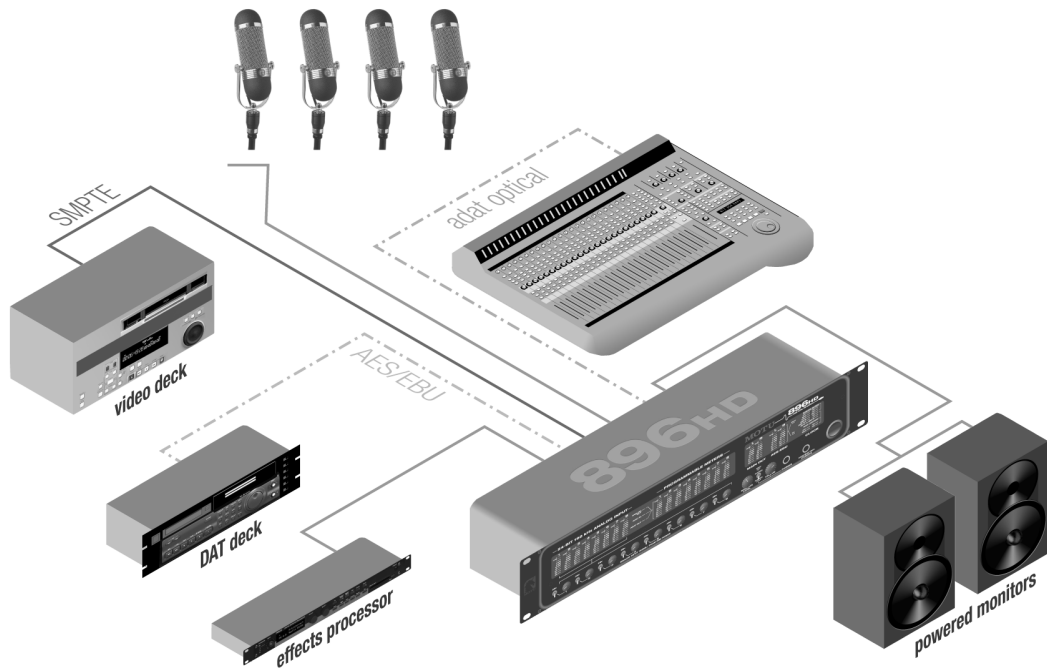


Figure 9-3: An example setup of a system that takes full advantage of CueMix DSP.

STAND-ALONE OPERATION

All settings, including all mix settings and global settings, are saved in the 896HD's memory, and they remain in effect even when the 896HD is not connected to a computer. This allows you to use the 896HD as a stand-alone 8-bus mixer. You can make adjustments to the four mix bus master faders at any time from the front panel.

CUEMIX CONSOLE EXAMPLES

Figure 9-3 shows some examples of how you can use CueMix DSP:

- Powered speakers are connected to the 896HD main outs. Any input can be routed directly to the speakers.
- Microphone input can be routed via CueMix DSP to the effects processor for live outboard processing during recording. The resulting signal

can be recorded into the computer either wet, dry or both (via the effects processor return or the direct mic input).

- The ADAT optical connection provides 8 channels of 24-bit digital I/O to the digital mixer (or 4 channels at 96kHz). Any device connected to the 896HD can be routed to/from the mixer with no latency. Conversely, any mixer channel can be routed to any device connected to the 896HD with no latency.
- Another example of ADAT optical connectivity is to use Giga Studio, and use CueMix DSP to route Giga inputs directly to the powered monitors connected to the 896HD for no-latency monitoring of your Giga tracks.

CHAPTER 10 Performance Tips & Troubleshooting

Why does the start sound not play through the 896HD?

The 896HD will only playback audio at a sampling rate of 44.1, 48, 88.2, 96, 176.4 or 192kHz.

The computer freezes when it starts up

If the computer is unable to boot up, it may be a conflict with the WDM Driver. As soon as the computer boots up, Windows will try and initialize the WDM Driver. If this fails, your computer will hang. To determine if the WDM Driver is the problem, boot up in Safe Mode or remove the MOTU Audio software with the Add/Remove Programs Control Panel then restart. Reinstall the 896HD software and choose only to install the MOTU FireWire ASIO driver, not the WDM Driver. Restart again.

'New hardware detected' window

If you connect your 896HD before running the 896HD Software Installer CD, Windows will prompt you that new hardware has been detected. Cancel this, and run the 896HD Software Installer, rather than allowing Windows to locate the drivers.

Audio in SONAR speeds up or slows down

This could be caused by having incorrect SMPTE/MTC Sync settings when you record audio into Sonar. These settings are located in the *Advanced* Tab in the Audio Options window, accessed from Sonar's Options menu. Make sure that you have the SMPTE/MTC Sync set to "Trigger and Freewheel" before recording audio while slaved to external time code.

896HD inputs and outputs are not available in SONAR

Make sure that the inputs and/or outputs that you want to use are enabled in SONAR. Also run the Wave Profiler in the Audio preferences dialog.

896HD inputs and outputs are not available in Cubase

Make sure that the inputs and outputs are enabled in Cubase.

Sound Forge and 'Smooth Scroll'

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

No input from an ADAT

If you are having trouble recording on your ADAT from the 896HD, check the Digital input setting. After power cycling, tape decks often come up configured to record from their analog inputs. You won't be able to record from the 896HD to a tape deck until it is switched to digital input.

Clicks and pops under word clock sync

Many problems result from incorrect word clocking. It is essential that all digital devices in the system be word locked. Consult chapter 4, "Installing the 896HD Hardware" (page 17) for detailed information on how to word clock your gear. Whenever there is any weird noise or distortion, suspect incorrect word lock.

Clicks and pops under ADAT Sync

Sometimes, the ADAT sync cable seems to be plugged into the 896HD, and partially works - but it isn't really all the way in. This can cause clicks when slaved to ADAT 9-pin. Make sure the cable is seated firmly.

Clicks and pops due to hard drive problems

If you have checked your clock settings sync cable connections and you are still getting clicks and pops in your audio, you may have a drive related problem. Set the 896HD Clock Source to Internal and try recording just using the analog inputs and outputs of the 896HD. If you encounter the same artifacts you may want try using another drive in your computer. Clicks and pops can also occur when the drive is severely fragmented, the disk drivers are outdated, or if you are using a SCSI accelerator that is not optimally configured for working with audio.

Connecting or powering gear during operation

It is not recommended that you connect/disconnect, or power on/off devices connected to the 896HD while recording or playing back audio. Doing so may cause a brief glitch in the audio.

Monitoring - How to monitor inputs?

Please refer to the documentation for the audio application that you are using. If your application does not support input monitoring, you will need to use the 896HD's hardware-based CueMix DSP monitoring feature. Please see chapter 8, "Reducing Monitoring Latency" (page 59).

No optical inputs or outputs are available in host audio application

Check to make sure you have the desired optical inputs and/or outputs enabled in the MOTU FireWire Audio Console.

Controlling monitoring latency

See chapter 8, "Reducing Monitoring Latency" (page 59).

CUSTOMER SUPPORT

We are happy to provide customer support to our registered users. If you haven't already done so, please take a moment to complete the registration card included with your 896HD. When we receive

your card, you'll be placed on our mailing list for free software updates and information about new products.

REPLACING DISKS

If your 896HD installer CD becomes damaged and fails to provide you with fresh, working copies of the program, our Customer Support Department will be glad to replace it. You can request a replacement disc by calling our business office at (617) 576-2760 and asking for the customer service department.

TECHNICAL SUPPORT

If you are unable, with your dealer's help, to solve problems you encounter with the 896HD system, you may contact our technical support department in one of the following ways:

- Tech support hotline: (617) 576-3066 (Monday through Friday, 9 am to 6 pm EST)
- Tech support 24-hour fax line: (617) 354-3068
- Tech support email: techsupport@motu.com
- Web site: www.motu.com

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

- The serial number of the 896HD system. This is printed on a sticker placed on the bottom of the 896HD rack unit. You must be able to supply this number to receive technical support.
- Software version numbers for the audio software you are using, the MOTU FireWire Audio WDM Driver, MOTU FireWire ASIO driver, etc.
- A brief explanation of the problem, including the exact sequence of actions which cause it, and the contents of any error messages which appear on the screen.

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