

MOTU 896mk3 Hybrid™

User Guide for Windows

MOTU

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SAFETY PRECAUTIONS AND ELECTRICAL REQUIREMENTS

CAUTION! READ THIS SAFETY GUIDE BEFORE YOU BEGIN INSTALLATION OR OPERATION. FAILURE TO COMPLY WITH SAFETY INSTRUCTIONS COULD RESULT IN BODILY INJURY OR EQUIPMENT DAMAGE.

HAZARDOUS VOLAGES: CONTACT MAY CAUSE ELECTRIC SHOCK OR BURN. TURN OFF UNIT BEFORE SERVICING.

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 896mk3 COULD CAUSE AN ELECTRICAL SHOCK.

The MOTU 896mk3 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 896mk3 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

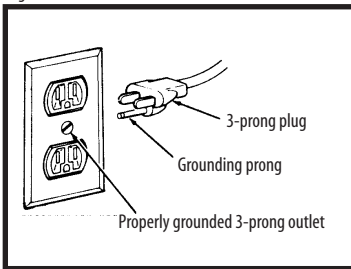
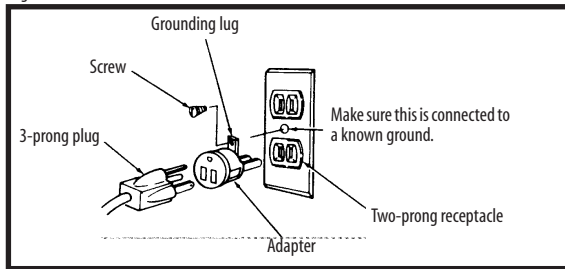


Figure B



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 896mk3 plug.

IMPORTANT SAFEGUARDS

1. Read these instructions. All the safety and operating instructions should be read before operating the 896mk3.
2. Keep these instructions. These safety instructions and the 896mk3 owner's manual should be retained for future reference.
3. Heed all warnings. All warnings on the 896mk3 and in the owner's manual should be adhered to.
4. Follow all Instructions. All operating and use instructions should be followed.
5. Do not use the 896mk3 near water.
6. Cleaning - Unplug the 896mk3 from the computer and clean only with a dry cloth. Do not use liquid or aerosol cleaners.
7. Ventilation - Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Heat - Do not install the 896mk3 near any heat sources such as radiators, heat registers, stoves, or another apparatus (including an amplifier) that produces heat.
9. Overloading - Do not overload wall outlets and extension cords as this can result in a risk of fire or electrical shock.
10. Grounding - Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding-type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
11. Power cord - Protect the 896mk3 power cord from being 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 unit.
12. Power switch - Install the 896mk3 so that the power switch can be accessed and operated at all times.
13. Disconnect - The main plug is considered to be the disconnect device for the 896mk3 and shall remain readily operable.
14. Accessories - Only use attachments/accessories specified by the manufacturer.
15. Placement - Use only with the cart, stand, tripod, bracket or table specified by the manufacturer, or sold with the 896mk3. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
16. Surge protection - Unplug the 896mk3 during lightning storms or when unused for long periods of time.
17. Servicing - Refer all servicing to qualified service personnel. Servicing is required when the 896mk3 has been damaged in any way, such as when a power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the 896mk3, the 896mk3 has been exposed to rain or moisture, does not operate normally, or has been dropped.
18. Power Sources - Refer to the manufacturer's operating instructions for power requirements. Be advised that different operating voltages may require the use of a different line cord and/or attachment plug.
19. Installation - Do not install the 896mk3 in an unventilated rack, or directly above heat-producing equipment such as power amplifiers. Observe the maximum ambient operating temperature listed below.
20. Power amplifiers - Never attach audio power amplifier outputs directly to any of the unit's connectors.
21. 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.
22. Safety Check - Upon completion of any service or repairs to this MOTU 896mk3, 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)

TO REDUCE THE RISK OF ELECTRICAL SHOCK OR 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. Do not expose this apparatus to rain or moisture. Do not place objects containing liquids on it.

AC INPUT

100 - 240VAC ~ 50 / 60Hz • 25 Watts.



CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN



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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.

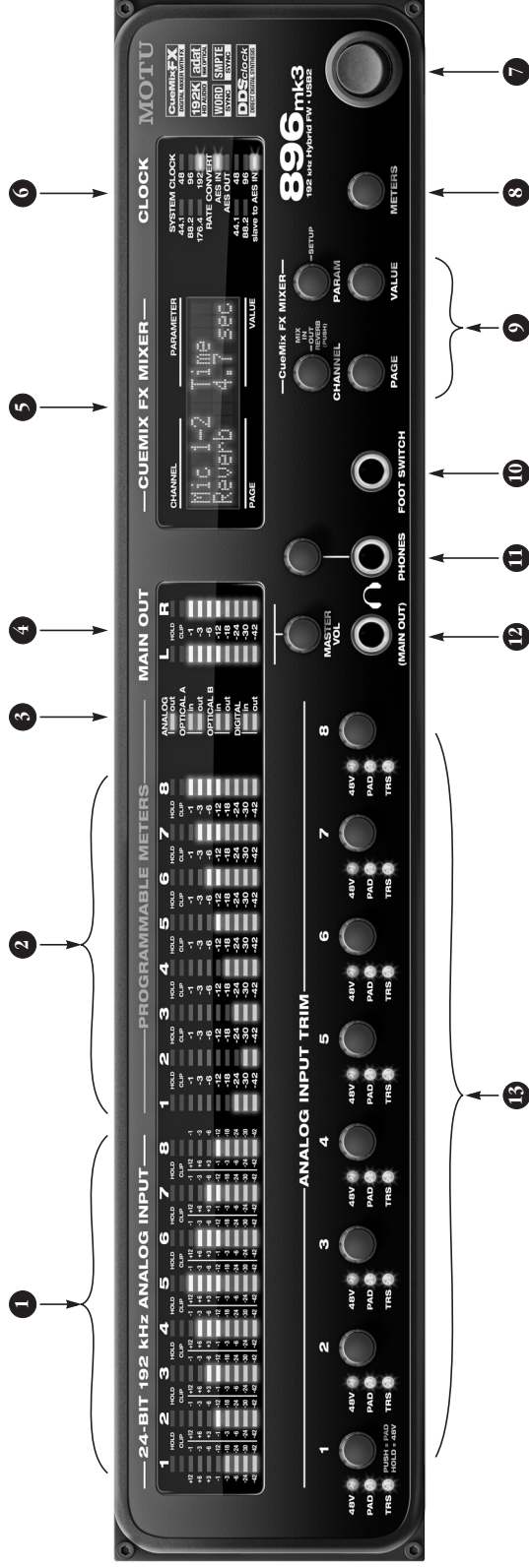
WARNING: changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.



Part 1

Getting Started

Quick Reference: 896mk3 Front Panel



- These 10-segment level meters are dedicated to the 896mk3's eight analog inputs. The top red 'HOLD' LED illuminates when the signal reaches full scale—for even just one sample—and remains illuminated until you clear it in the software. The 'CLIP' LED below only lights up momentarily so that you can continue to adjust level even after clipping has just occurred. The remaining LEDs provide two different scales for operation with or without V-Limit™, which provides hardware limiting and clip prevention for signal levels as high as +12dB over zero. Without V-Limit, the LED scale goes up to -1 dB and clips at zero. With V-Limit, the LED scale goes up to +12 dB and clips when the input signal goes over +12 dB.
- This bank of level meters can be programmed to display any input or output bank: analog, optical (ADAT or TOSLINK), or 'digital' (AES/EBU and S/PDIF). Use MOTU Audio Console or the METERS knob below to choose which bank you'd like to view with these meters. The LEDs to the right show which bank you are currently monitoring.
- The light that is illuminated here tells you which bank you are monitoring with the programmable meter bank to the left.
- These meters display the level for the 896mk3's main outputs.
- This section controls the 896mk3's built-in CueMix FX mixer and effects. There are eight stereo mix busses: each bus mixes all inputs (or any subset you wish) to a stereo output of your choice. You can apply EQ and compression to inputs and outputs, plus reverb to ins, outs and mixes. The four knobs below the LCD correspond directly to the four labeled sections of the LCD. Use the CHANNEL knob to choose the input, output or mix you wish to edit. Push it to switch among inputs, outputs and busses, then turn it to choose the desired channel or bus. Use the PAGE, PARAM and VALUE knobs to access the settings for the chosen channel.
- The CLOCK LEDs indicate the global sample rate at which the MOTU 896mk3 is operating. Use MOTU 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 896mk3 to slave to an external clock, such as ADAT, but there is no clock signal currently being detected, these lights will flash.
- FireWire and USB are "plug-and-play" protocols. That means that you can turn off the MOTU 896mk3 and turn it back on (or even unplug it) without restarting your computer.
- Use this knob to choose which audio bank you are viewing with the programmable meters (2).
- Push the CHANNEL knob repeatedly to cycle among the four main sections of the mixer: mix busses, inputs, outputs and the reverb module. Push the PARAM knob to enter the SETUP menu, which provides global 896mk3 settings, such as the global sample rate, etc.
- 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 43.
- From the factory, the PHONES jack is a discrete output (at 44.1/48 kHz), but it can mirror any other output pair (digital or analog). For example, at 88.2/96 kHz, it defaults to mirroring the main outputs. As the primary phone jack, it has its own dedicated volume knob.
- The phone jack labeled (MAIN OUT) is a standard quarter-inch stereo headphone jack. Its output is hard-wired to mirror the XLR main outs on the rear panel. From the factory, the MASTER VOL knob above it controls the main outputs and this jack, but MASTER VOL can be programmed to control any combination of outputs. See "The Monitor Group" on page 95 for details. Push the knob once to view the current volume setting in the LCD display; push it again to mute the monitor group; push a third time to return to the previous volume. Note: if the Monitor group is programmed to not include the main outputs, the MASTER VOL knob will no longer control the volume of this phone jack, either.
- These eight trim knobs provide approximately 53 dB of gain for the XLR mic inputs and the hi-Z IHS guitar/instrument inputs on the rear panel. All eight inputs have preamps, so you can plug just about anything into them: a microphone, a guitar, or a synth. If you plug in a +4 or -10 line level signal, use the quarter-inch jack and be sure to enable the -20 dB pad. Use the trim knobs and the "ANALOG INPUT" meters in the metering section above to calibrate the input signal level. The meters cover both the IHS and XLR input.
- Push the TRIM encoder to toggle a -20 dB pad; push and hold to toggle 48V phantom power.

These mic inputs are also equipped with the 896mk3's V-Limit™ hardware limiter, which provides an additional +12 dB of headroom above zero with no clipping or digital distortion. See "Mic/guitar inputs meters with V-Limit™ compressor" on page 42 for details.

Quick Reference: 896mk3 Rear Panel



1. 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 896mk3. For details, see "Syncing AES/EBU devices" on page 30. At the 4x sample rates, (176.4 and 192kHz), AES/EBU is disabled.
2. These two XLR jacks serve as the MOTU 896mk3's main analog outputs. You can connect them to a set of powered studio monitors and then control the volume from the front panel MASTER VOL knob. These jacks are always mirrored by the MAIN OUT headphone jack on the front panel.

To hear disk tracks in your audio software on these main outs, assign the disk tracks (and master fader) to these main outs. Also make sure the Main Out Assign option is set to *Main Out 1-2*. See "Main Out Assign" on page 44. You can also use CueMix FX to monitor live 896mk3 inputs here as well.
3. The 896mk3's eight analog outputs are XLR connectors referenced to +19 dBu. They are equipped with 24-bit D/A converters capable of 192kHz.
4. These XLR/TPS combo jacks accept either a mic cable or a quarter-inch cable. The XLR jack is equipped with a mic preamp. 48V phantom power and 20 dB pad can be applied via the trim knobs on the front panel. The trim knob provides 53 dB of gain. The quarter-inch jack can accept a guitar or -10/+4 dB line level input. When connecting a line level input, be sure to turn off phantom power first, and engage the -20 dB pad.
5. Connect the 896mk3 to the computer here via either FireWire or USB2, using the standard 1394 FireWire B or USB cable provided with your MOTU 896mk3. If you use FireWire, you can use the extra FireWire port to daisy-chain up to four MOTU FireWire audio interfaces to a single FireWire bus, or to connect other FireWire devices. Keep in mind that the 896mk3 uses more
6. These optical digital I/O connectors can be connected either to an ADAT-compatible "lightpipe" device (such as a digital mixer) or to a S/PDIF optical ("TOSLINK") compatible device such as an effects processor or DAT machine. Be sure to set the format in the MOTU Audio Console software (or using the front panel LCD). (See "Optical input/output" on page 44 for details.) ADAT optical supplies eight channels of 24-bit digital I/O per bank (4 channels per bank at 96kHz). TOSLINK is stereo at sample rates up to 96 kHz.
7. These jacks provide stereo, 24-bit S/PDIF digital input and output at all supported sample rates (up to 96 kHz).
8. 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 896mk3.
9. The 896mk3 is equipped with an auto-switching international power supply.

FireWire bus bandwidth when one or both optical banks are enabled, or when it operates at higher sample rates. These operating configurations will limit the number of devices you can daisy chain on a single FireWire bus. For details, see "Connecting multiple MOTU FireWire interfaces" on page 35.

One special note: you can choose independent formats for each bank, A and B, as well as IN and OUT within each bank. For example, you could choose ADAT for the optical A IN (for, say, eight channels of input from your digital mixer) and stereo TOSLINK for the optical A OUT (for, say, your DAT machine).

Quick Reference: MOTU Audio Console

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 42.

Uncheck this option if the Windows audio software you are using with the 896mk3 does not support Windows WaveRT drivers and instead only supports WDM drivers.

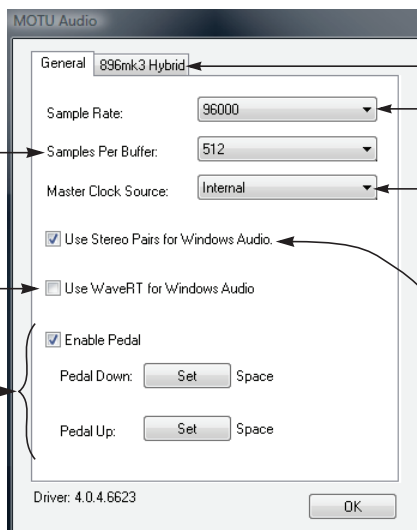
If you have a foot switch connected to the 896mk3, these settings let you map the foot switch to any computer keyboard key for both the up and down position. For details about how to set this up, see “Enable Pedal” on page 43.

This menu lets you choose what you will hear from the PHONES jack. To mirror the main outs, choose *Main Out 1-2*. Or you can mirror any other output pair. To hear the phones as their own independent output, choose *Phones 1-2* (at 44.1 or 48 kHz). At higher sample rates, the phones must mirror any other available output pair.)

Choose the output pair you would like the main outs to mirror, or choose *Main Out 1-2* to operate them as their own independent pair.

The 896mk3 driver provides a stereo return back to the computer. This return feeds the signal on any 896mk3 output pair directly back to the computer, where you can record, process, monitor or otherwise use it. This is a great way to “bounce” full mixes, complete with live audio routed through the 896mk3 only, back into the computer.

If you are running an 896mk3 interface at a high sample rate (96, 88.2, 176.4 or 192 kHz), this option appears in the interface tab. It lets you choose a word clock output rate that either matches the global sample rate (e.g. 96kHz) or reduces it to the corresponding 1x rate (e.g. 48kHz instead of 192 kHz).

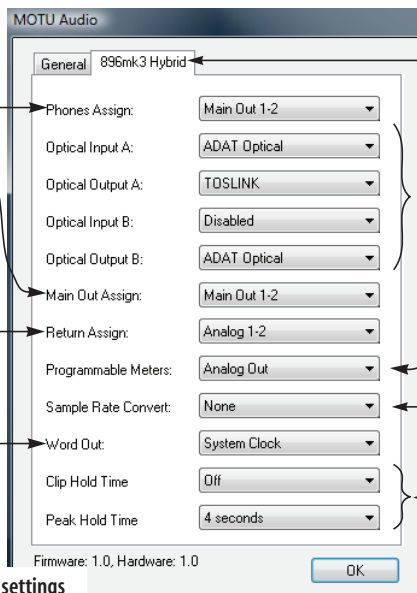


Click the tabs to access general MOTU interface settings or settings specific to the 896mk3 (or other connected interface.)

Choose the global sample rate for the system here.

Determines the clock source for your 896mk3. If you're just using the analog ins and outs, set this to 'Internal'. The other settings are for digital transfers via AES/EBU, S/PDIF or optical ports, or for slaving the 896mk3 to word clock.

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 896mk3 tab to access these settings.

Each optical bank can be configured independently: ADAT or TOSLINK. Disable them when not in use to conserve DSP and FireWire/USB bus bandwidth.

Lets you choose what to monitor with the 896mk3's programmable front panel meters.

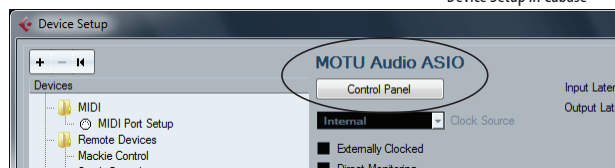
Provides several options for the 896mk3's AES/EBU sample rate conversion. See, “Syncing AES/EBU devices” on page 30.

The *Clip Hold Time* controls how long the top-most red LED remains illuminated after clipping. The *Peak Hold Time* controls how long the highest illuminated LED remains lit before going dark.

How to access these settings

There are several ways to access these settings:

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



Device Setup in Cubase



CHAPTER 1 **About the 896mk3 Hybrid**

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OVERVIEW

The 896mk3 Hybrid is a hybrid FireWire USB2 audio interface for Mac and Windows that provides 28 separate inputs and 32 separate outputs (at 44.1 or 48kHz). Both analog and digital I/O are offered at sample rates up to 96 kHz, and analog recording and playback is offered at rates up to 192 kHz. All inputs and outputs can be accessed simultaneously. The 896mk3 Hybrid consists of a standard 19-inch, double space, rack-mountable I/O unit that connects directly to a computer via a standard FireWire or USB cable.

The 896mk3 offers the following main features:

- Universal computer connectivity via FireWire or high-speed USB2
- Eight 24-bit XLR analog outputs
- Eight 24-bit analog inputs equipped with XLR/TRS “combo” jacks
- Eight mic preamps (one on each input)
- Independent 48V phantom power and 20 dB pad for each mic input
- Independent front-panel Precision Digital Trim™ for each input
- Two extra XLR analog main outs
- Operation on all analog I/O at standard sample rates up to 192 kHz
- Two banks of optical digital I/O that provide 16 channels of ADAT optical at 48 kHz, 8 channels of S/MUX optical I/O at 96 kHz or two banks of stereo TOSLINK at rates up to 96 kHz
- AES/EBU digital I/O at samples rates up to 96 kHz with sample rate conversion
- RCA S/PDIF at sample rates up to 96 kHz
- Word clock input and output
- On-board SMPTE synchronization via any analog input and output
- Two FireWire jacks for chaining multiple units
- Foot switch input for hands-free punch-in/out
- Two phone jacks with independent volume control
- Programmable master volume knob
- CueMix™ FX no-latency mixing, monitoring and effects processing
- Front-panel LCD programming for the mixer and all other settings
- Dedicated 10-segment LED level meters for all eight analog inputs
- Dedicated 10-segment level meters for main outs
- Programmable 10-segment LED level meters that display levels for any 8-channel input or output bank, as well as AES/EBU or S/PDIF digital I/O
- Auto-switching international power supply
- Stand-alone operation
- Mac and Windows drivers for multi-channel operation and across-the-board compatibility with any audio software on current Mac and Windows systems

With a variety of I/O formats, mic preamps, no-latency mixing and processing of live input and synchronization capabilities, the 896mk3 Hybrid is a complete, portable “studio in a box” when used with a Mac or Windows computer.

THE 896MK3 HYBRID REAR PANEL

The 896mk3 Hybrid rear panel has the following connectors:

- Eight 24-bit 192 kHz XLR analog outputs
- Eight 24-bit 192 kHz “combo” (XLR + balanced quarter-inch) analog inputs, each equipped with a mic preamp, front-panel 48V phantom power switch, pad switch, and trim knob
- Two XLR main analog outputs with front-panel volume knob
- Two sets of optical connectors (in and out), individually switchable among ADAT optical “lightpipe”, 96 kHz S/MUX optical or S/PDIF “TOSLINK”
- AES/EBU input and output
- RCA S/PDIF in/out
- BNC word clock input and output
- Two 1394 FireWire B connectors
- One high-speed USB2 connector

28 inputs and 32 outputs

All 896mk3 inputs and outputs can be used simultaneously, for a total of 28 inputs and 32 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†	16	16
AES/EBU 24-bit 96kHz digital	stereo	stereo
RCA S/PDIF 24-bit 96kHz digital	stereo	stereo
Total	28	32

* The phone jack below the MASTER VOL knob is hard-wired to (mirrors) the XLR main outs. The PHONES output can operate as an independent output pair, or it can mirror any other 896mk3 output pair, such as the main outs.

† The 896mk3 optical connectors support several standard optical I/O formats, which provide varying channel counts. See “Optical input/output” on page 44 for details about optical bank operation.

With the exception of the phone jack on the front panel labeled “(MAIN OUT)”, all inputs and outputs are discrete. For example, 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.

Analog

All 10 analog inputs and outputs are equipped with 24-bit 192 kHz A/D converters. All audio is carried to the computer in a 24-bit data stream.

Mic/guitar preamps

All eight analog inputs are equipped with a preamp on a combo-style connector that accepts either an XLR or quarter-inch (guitar) plug. Individual 48V

phantom power and 20 dB pad are supplied by separate front panel switches for each input. In addition, each input has its own trim knob.

Precision Digital Trim™

All of the 896mk3's analog inputs are equipped with digitally controlled analog trims, adjustable in approximately 1 dB increments. The mic/instrument input trims can be adjusted using front-panel digital rotary encoders that provide feedback in the front panel LCD with up to 53 dB of boost. All analog inputs can be trimmed with the front panel knobs or with the 896mk3's included CueMix FX control software for Mac and Windows. This gives you finely-tuned control of trim settings for a wide variety of analog inputs for optimum levels. Different trim configurations can then be saved as preset configurations for instant recall.

Analog input overload protection

All eight mic inputs are equipped with V-Limit™, a hardware limiter that helps prevent digital clipping from overloaded input signals. With V-Limit enabled, signals can go above zero dB (with limiting applied) to as high as +12 dB above zero with no distortion due to digital clipping.

Additional or alternative protection can be applied to the mic/guitar inputs by enabling the 896mk3's Soft Clip feature, which engages just before clipping occurs and helps reduce perceptible distortion.

Main Outs

The main outs are equipped with 24-bit 192kHz D/A converters and serve as independent outputs for the computer or for the 896mk3's on-board CueMix FX mixes. The main out volume can be controlled with the front panel volume knob.

Optical

The two optical banks provide 16 channels of ADAT optical at 44.1 or 48 kHz, 8 channels of S/MUX optical I/O at 96 kHz or two banks of stereo TOSLINK at rates up to 96 kHz. The banks operate independently, including input and output, allowing you to mix and match any optical formats. For example, you could receive 4 channels of 96 kHz S/MUX input on Bank A while at the same time sending 96 kHz stereo optical S/PDIF ("TOSLINK") from the Bank A output.

AES/EBU with sample rate conversion

The 896mk3 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 896mk3's global sample rate. The AES/EBU jacks are disabled at the 4x sample rates (176.4 and 192kHz).

S/PDIF

The 896mk3 rear panel provides S/PDIF input and output in two different formats: RCA "coax" and optical "TOSLINK". The RCA jacks are dedicated to the S/PDIF format. The TOSLINK jacks can be used for either TOSLINK or ADAT optical, as discussed earlier.

Word clock

The 896mk3 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 896mk3 global sample rate is set to 96 kHz, the word clock input can resolve to a "low" rate of 48 kHz. Similarly, when the 896mk3 is operating at 192 kHz, MOTU Audio Console lets you choose a word clock output rate of 48 kHz (the *Force 44.1/48kHz* setting).

On-board SMPTE synchronization

The 896mk3 can resolve directly to SMPTE time code via any TRS analog input, without a separate synchronizer. The 896mk3 can also generate SMPTE time code via any XLR analog output. The 896mk3 provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy.

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

Hybrid FireWire/USB2 connectivity

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

To fully support both formats, the 896mk3 Hybrid audio interface is equipped with both FireWire B (400 Mbit/sec) connectors and a high-speed USB2 (480 Mbit/sec) connector, and you can use either port to connect the 896mk3 to your computer. This gives you maximum flexibility and compatibility with today's ever-expanding universe of Mac and Windows computers.

THE 896MK3 HYBRID FRONT PANEL

Metering

The front panel of the MOTU 896mk3 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 any other bank

you choose with the Meters knob: Analog out, optical I/O (ADAT or TOSLINK) or digital I/O (AES/EBU and S/PDIF). A status LED to the right shows which bank you are currently viewing. You can also adjust this setting in the MOTU Audio Console software.

The 896mk3 front panel also displays stereo meters for the main analog outs.

The *Clock* lights indicate the global sample rate (as chosen in the MOTU Audio Console software). The Rate Convert LEDs indicate if sample rate conversion is being applied to the AES/EBU input or output, and if so, the rate being converted to.

Input trim knobs, phantom power & 20 dB Pad

The front-panel input trim knobs provide 53 dB of boost, -20 dB pad (by pushing the knob), and 48V phantom power (by holding in the knob) for each mic input.

Headphone output and main volume control

The 896mk3 front panel provides two independent headphone jacks with independent volume knobs, one of which also controls the XLR main outs on the rear panel. Alternately, this MASTER VOL knob can be programmed to control any combination of outputs (analog and/or digital). For example, it can control monitor output for an entire 5.1 or 7.1 surround mix.

Foot switch

The quarter-inch Foot Switch jack accepts a standard foot switch. When you push the foot switch, the 896mk3 triggers a programmable keystroke on the computer keyboard. For example, you could program the foot switch to toggle recording in your host software. The MOTU Audio Console software lets you program any keystroke you wish.

Programmable backlit LCD display

Any 896mk3 setting, including the powerful CueMix FX on-board 16-bus mixer with effects, can be accessed directly from the front panel using the four rotary encoders and the 2x16 backlit LCD display.

16-BIT AND 24-BIT RECORDING

The 896mk3 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 896mk3's analog or digital inputs and outputs. 24-bit audio files can be recorded with any compatible host application that supports 24-bit recording.

CUEMIX FX 32-BIT FLOATING POINT MIXING AND EFFECTS

All 896mk3 inputs and outputs can be routed to the on-board CueMix FX 16-bus (8 stereo) digital mixer driven by hardware-based DSP with 32-bit floating point precision. The mixer allows you to apply no-latency effects processing to inputs, outputs or busses directly in the 896mk3 hardware, independent of the computer. Effects can even be applied when the 896mk3 is operating stand-alone (without a computer) as a complete rack-mounted mixer. Input signals to the computer can be recorded wet, dry, or dry with a wet monitor mix (for musicians during recording, for example).

Effects include reverb, parametric EQ and compression/limiting. The 896mk3's Classic Reverb™ provides five different room types, three frequency bands with adjustable crossover points, shelf filtering and reverb lengths up to 60-seconds.

Two forms of compression are supplied: a standard compressor with conventional threshold/ratio/attack/release/gain controls and the Leveler™, an accurate model of the legendary LA-2A optical compressor, which provides vintage, musical automatic gain control.

CueMix FX also provides 7-band parametric EQ modeled after British analog console EQs, featuring 4 filter styles (gain/Q profiles) to effectively cover a wide range of audio material. Low-pass and high-pass filters are also supplied with slopes that range from 6 to 36 dB. The EQ employs extremely high precision 64-bit floating point processing.

The 896mk3's flexible effects architecture allows you to apply EQ and compression on every input and output (a total of 58 channels), with enough DSP resources for at least one band of parametric EQ and compression on every channel at 48 kHz. However, DSP resources are allocated dynamically and a DSP meter in the CueMix FX software allows you to keep tabs on the 896mk3's processing resources. Each input, output and mix bus provides a send to the Classic Reverb processor, which then feeds reverb returns to mix busses and outputs, with a selectable split point between them to prevent send/return feedback loops.

INCLUDED SOFTWARE

The 896mk3 software installer provides the following including cross-platform software applications for Mac and Windows.

MOTU Audio Console

MOTU Audio Console provides access to basic hardware settings, such as sample rate, optical I/O format selection, headphone output channel selection, foot switch input programming, and other settings.

MOTU SMPTE Console

MOTU SMPTE Console provides access to the 896mk3 system's SMPTE time code synchronization features, including locking to time code, generating time code, displaying a time code read-out, and so on.

CueMix FX

CueMix FX gives you complete control over the 896mk3's CueMix FX on-board mixer, which provides no-latency monitoring, mixing and processing of live inputs through your 896mk3.

CueMix FX provides attractive graphic mixing, graphic editing of parametric EQ and a convenient tabbed interface for quick access to all mixing features, digitally controlled trims and other settings in your MOTU audio interface.

CueMix FX provides many advanced features, such as an accurate instrument tuner and an extensive arsenal of audio analysis tools, including a real-time FFT, spectrogram “waterfall” display, oscilloscope, and phase analysis tools.

HOST AUDIO SOFTWARE

The 896mk3 system ships with Windows drivers that allows you to record, edit, play back and mix your 896mk3 projects using your favorite Windows audio software.

CHAPTER 2 **Packing List and System Requirements**

PACKING LIST

The 896mk3 Hybrid ships with the items listed below. If any of these items are not present in your 896mk3 box when you first open it, please immediately contact your dealer or MOTU.

- One 896mk3 Hybrid I/O rack unit
- One 9-pin to 9-pin IEEE 1394 “FireWire” B cable
- One USB cable
- Power cord
- One 896mk3 Hybrid Mac/Windows manual
- One cross-platform installer disc
- Product registration card

WINDOWS SYSTEM REQUIREMENTS

The 896mk3 system requires the following Windows system:

- A 1 GHz Pentium-based PC compatible or faster equipped with at least one USB2 or FireWire port
- 1 GB RAM; 2 GB or more recommended
- Windows 7 or Vista, 32- or 64-bit; Vista SP 2 or later required
- Available FireWire or USB2 port
- A large hard drive (preferably at least 250 GB)

PLEASE REGISTER TODAY!

Please register your 896mk3 today. There are two ways to register.

- Visit www.motu.com/register

OR

- Fill out and mail the included product registration card

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

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

CHAPTER 3 Installing the 896mk3 Software

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MOTU Audio Console..... 20

CueMix FX 20

MOTU SMPTE Console 20

INSTALLATION

Before you connect the 896mk3 to your computer and power it on, run the 896mk3 software installer. This ensures that all the 896mk3 components are properly installed in your system.

☛ If you’ve already connected the 896mk3 to your computer and powered it on, Windows may issue an alert notifying you that the 896mk3 requires drivers, followed by another window asking you to locate the drivers. If this happens:

- 1 Cancel the driver search.

- 2 Power off and disconnect the 896mk3.

- 3 Run the MOTU Audio Installer as instructed in the next section.

Run the MOTU Audio installer

Install the 896mk3 Hybrid software as follows:

- 1 Insert the MOTU Audio Installer disc; or, if you have downloaded the MOTU Audio installer, locate the folder containing the download.
- 2 Read the *Read Me* file for installation assistance and other important information.
- 3 Open the *Setup Audio* application.
- 4 Follow the directions that the installer gives you.

Drivers are installed, along with MOTU Audio Console, CueMix FX, and other components, summarized in the table below.

Software component	Purpose	For more information
MOTU Audio and MIDI drivers	Provides multi-channel audio input and output for MOTU FireWire and USB Audio devices with host audio software.	“MOTU Audio drivers” on page 20
MOTU Audio Console	Provides access to all of the settings in the 896mk3 and other MOTU interfaces. Required for 896mk3 operation.	chapter 5, “MOTU Audio Console” (page 39)
CueMix FX	Gives you complete control over the 896mk3’s CueMix FX on-board mixer, which provides no-latency monitoring, mixing and processing of live inputs through your 896mk3.	chapter 9, “CueMix FX” (page 71)
MOTU SMPTE Console	Provides access to the 896mk3 system’s SMPTE time code sync features.	chapter 10, “MOTU SMPTE Console” (page 119)

MOTU AUDIO DRIVERS

ASIO

ASIO is an acronym for *Audio Streaming Input and Output*. The MOTU Audio ASIO driver provides multi-channel audio input and output for applications that support ASIO audio drivers, such as Ableton Live, Avid Pro Tools, Cakewalk SONAR, Cockos Reaper, Propellerhead Reason and Record, Steinberg Cubase and Nuendo, and others.

For details about using the 896mk3 with ASIO, see chapter 7, “Configuring Host Audio Software” (page 57).

WDM

WDM is an acronym for *Windows Driver Model*. The MOTU Audio WDM driver provides multi-channel audio input and output for applications that support WDM audio drivers.

For details about using the 896mk3 with WDM, see chapter 7, “Configuring Host Audio Software” (page 57).

WaveRT

Uncheck this option if the Windows audio software you are using with your MOTU audio interface does not support Windows WaveRT drivers and instead only supports WDM drivers.

MOTU AUDIO CONSOLE

MOTU Audio Console (available in the Start menu) gives you access to all of the settings in the 896mk3, such as the clock source and sample rate. For complete details, see chapter 5, “MOTU Audio Console” (page 39).

CUEMIX FX

CueMix FX (available in the Start menu) provides control over the 896mk3’s no-latency CueMix FX on-board mixing and effects processing. For details, see chapter 9, “CueMix FX” (page 71).

MOTU SMPTE CONSOLE

MOTU SMPTE Console (available in the Start menu) software provides a complete set of tools to resolve the 896mk3 to SMPTE time code, and to generate SMPTE for striping, regenerating or slaving other devices to the computer. For details, see chapter 10, “MOTU SMPTE Console” (page 119).

CHAPTER 4 Installing the 896mk3 Hardware

OVERVIEW

Here's an overview for installing the 896mk3:

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Connect the 896mk3 to the computer.	
Connect audio inputs and outputs	23
Make optical and analog connections as desired.	
Connect a foot switch.....	25
Connect a footswitch to trigger any keystroke.	
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CONNECT THE 896MK3 INTERFACE

Your 896mk3 Hybrid audio interface is equipped with both FireWire B connectors (running at 400 Mbit/sec) and a high-speed USB2 connector (480 Mbit/sec), and you can use either port to connect the 896mk3 to your computer. This gives you maximum flexibility and compatibility with today's ever-expanding universe of Mac and Windows computers.

Type B FireWire ports

The 896mk3 Hybrid has two FireWire Type B ports, which provide the most reliable FireWire connection available. The ports operate at 400 Mbit/s, and they can be connected to any available FireWire port on your computer, either Type A or Type B. If your computer has FireWire Type B ports, use the included 9-pin-to-9-pin FireWire cable. If your computer has either standard Type A ports or miniature Type A ports, use the appropriate 9-pin-to-6-pin or 9-pin-to-4-pin FireWire cable (sold separately).

Which should I use: FireWire or USB2?

If your computer does not have a FireWire port, then obviously you will need to connect the 896mk3 Hybrid to one of its high-speed USB 2.0 ports.

If your computer has both FireWire and USB2, then it is your choice, and your decision may depend mostly on other peripherals you may have.

If you are connecting via FireWire

- 1 Before you begin, make sure your computer and the 896mk3 are switched off.
- 2 Plug one end of the 896mk3 FireWire cable (included) into the FireWire socket on the computer as shown below in Figure 4-1.
 You can also connect the 896mk3 to a 400Mbit "FireWire A" port using a 9-pin-to-6-pin FireWire B cable (not included). The 896mk3 will still operate at its specified 400Mbit (FireWire A) data rate.
- 3 Plug the other end of the FireWire cable into the 896mk3 as shown below in Figure 4-1.



Figure 4-1: Connecting the 896mk3 to the computer via FireWire.

☞ Make absolute sure to align the notched side of the FireWire plug properly with the notched side of the FireWire socket on the 896mk3. If you attempt to force the plug into the socket the wrong way, you can damage the 896mk3.

High Speed USB 2.0 versus USB 1.1

There are primarily two types of USB host controllers widely available on current personal computers. USB 1.1 controllers support simple peripherals that don't require a high speed connection, such as a computer keyboard, a mouse, or a printer. USB 2.0 controllers support high speed devices such as the 896mk3. Since the 896mk3 requires a high speed connection, it must be connected to a USB 2.0 host controller or hub.

For the most reliable connection, it is recommended that you connect the 896mk3 directly to one of your computer's USB 2.0-compatible ports. However, since USB 2.0 hubs are compatible with both types of devices, the 896mk3 can be connected to a USB 2.0 hub along with USB 1.1 devices if necessary. The 896mk3 will not operate properly if it is connected to a USB 1.1 hub.

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

- 1** Open the Windows Control Panel, and choose Hardware and Sound.
- 2** In the Devices and Printers section, click Device Manager.
- 3** In Device Manager, open the Universal Serial Bus Controllers section.

4 Look in the list of USB devices. An Enhanced USB Host Controller Interface (EHCI) represents a USB 2.0 controller. An Open USB Host Controller Interface (OHCI) or Universal USB Host Controller Interface (UHCI) represents a USB 1.1 controller.

If you are connecting via high-speed USB 2.0

- 1** Before you begin, make sure your computer and the 896mk3 are switched off.
- 2** Plug the flat “type A” plug of the 896mk3 USB cable (included) into a USB2-equipped socket on the computer as shown below in Figure 4-2.
- 3** Plug the squared “type B” plug of the USB cable into the 896mk3 I/O as shown below in Figure 4-2.



Figure 4-2: Connecting the 896mk3 to the computer via USB.

CONNECT AUDIO INPUTS AND OUTPUTS

The 896mk3 audio interface has the following audio input and output connectors:

- 8 combo-style XLR/quarter-inch analog inputs
- 8 XLR analog outputs
- 2 XLR main outs
- AES/EBU input/output
- S/PDIF input/output
- ADAT optical input/output

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

Analog inputs

Connect a microphone, guitar, instrument or line level analog input to any XLR/quarter-inch combo jack with either a standard mic cable or a balanced/unbalanced cable with a quarter-inch plug.

Phantom power

If you are connecting a condenser microphone or other device that requires phantom power, push and hold the corresponding front-panel Trim rotary encoder for a few seconds to toggle phantom power. The red LED will turn on or off accordingly.

☛ Always disable phantom power before inserting a quarter-inch plug.

Trim

Both the XLR mic input and the high-impedance quarter-inch guitar input are equipped with 53 dB gain. Use the corresponding front panel trim knob to adjust the input level as needed for each input. The LCD provides visual feedback as you turn the trim knob.

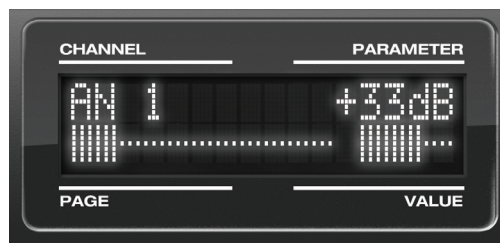


Figure 4-4: The LCD gives you feedback as you turn the TRIM knobs for the analog inputs.

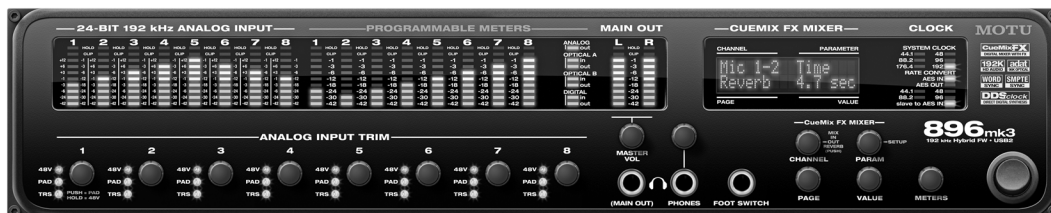
The 896mk3's input trims are digitally controlled, so they allow you to make fine-tuned adjustments in 1 dB increments. You can also adjust trim in the MOTU CueMix FX software. See “Input trim” on page 80.

-20 dB pad

If the input signal is too hot with the trim turned all the way down, engage the -20dB pad for the input on the front panel. To toggle the -20 dB pad for an input, quickly push its TRIM rotary encoder. The green LED will turn on or off accordingly.

☛ For line level quarter-inch inputs, turn on the pad.

Figure 4-3: the 896mk3 front panel.



Combo jack summary

Use these general guidelines for the 48V phantom power, pad and trim settings for the combo inputs:

Input	48V	Pad	Trim
Condenser mic	On	As needed	As needed
Dynamic mic	Off	As needed	As needed
Guitar	Off	Off	As needed
-10 dB Line level via TRS	Off	On	As needed
+4 dB Line Level via TRS	Off	On	As needed

Analog outputs

For the XLR outputs, use high-quality shielded cables.

Main outs

The XLR main outputs serve as independent outputs. From the factory, the main out volume is controlled by the MASTER VOL knob on the front panel, although this knob can be programmed to control any combination of outputs. For details, see “The Monitor Group” on page 95. In a standard studio configuration, the main outs are intended for a pair of studio monitors, but they can also be used as additional outputs for any purpose.

Optical

The 896mk3 rear panel provides two sets of ADAT optical (“lightpipe”) connectors: Bank A and B (Figure 4-5). Each bank provides an input and output connector. All four connectors can operate

independently and offer two different optical formats: ADAT optical or TOSLINK (optical S/PDIF). For example, you could connect 8-channel ADAT optical input from your digital mixer and stereo TOSLINK output to an effects processor.

The 896mk3 supplies +12dB of digital trim (boost) for each optical input, which can be adjusted from CueMix FX (“Input trim” on page 78) or the front panel (“The IN (inputs) menu” on page 52).

Below is a summary of optical formats:

Format	44.1 or 48 kHz	88.2 or 96 kHz
ADAT optical	8 channels	4 channels
TOSLINK	stereo	stereo

Optical operation at 44.1 or 48 kHz

When configured for ADAT “lightpipe”, an optical connector provides 8 channels at 44.1 and 48 kHz.

ADAT optical operation at 88.2 or 96 kHz

When configured for ADAT “lightpipe”, an optical connector provides four channels at 88.2 or 96 kHz (2x sample rates). When using the ADAT lightpipe format at a 2x rate, be sure to choose either Type I or Type II operation, as explained in “ADAT SMUX Type” on page 51.

Figure 4-5: the 896mk3 rear panel.



Using optical I/O to operate the 896mk3 as an 8-channel expander

When the 896mk3 is not connected directly to a computer via FireWire or USB, the optical outputs can be programmed (via the CueMix FX mixer) to mirror the incoming signal on any combination of the 896mk3's inputs. By connecting the 896mk3 optical outputs to another device, such as another ADAT-optical equipped interface or a digital mixer, you add additional inputs to your system.

To learn how to program the 896mk3 when it is operating as a stand-alone expander in this fashion, see chapter 6, "Front Panel Operation" (page 43).

Choosing a clock source for optical connections

Be sure to review the digital audio clocking issues, as explained in "Syncing optical devices" on page 30.

S/PDIF

If you make a S/PDIF digital audio connection to another device, be sure to review the digital audio clocking issues, as explained in "Syncing S/PDIF devices" on page 29.

The 896mk3 supplies +12dB of digital trim (boost) for the S/PDIF input pair, which can be adjusted from CueMix FX ("Input trim" on page 78) or the front panel ("The IN (inputs) menu" on page 52).

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.

Be sure to review the digital audio clocking issues, as explained in "Syncing AES/EBU devices" on page 30.

The 896mk3 supplies +12dB of digital trim (boost) for the AES/EBU input pair, which can be adjusted from CueMix FX ("Input trim" on page 78) or the front panel ("The IN (inputs) menu" on page 52).

CONNECT A FOOT SWITCH

If you would like to use a foot switch with your 896mk3, connect it to the FOOT SWITCH jack. See "Enable Pedal" on page 43 for information about how to program the foot switch to trigger any computer keystroke you wish.

A TYPICAL 896MK3 SETUP

Here is a typical 896mk3 studio setup. This rig can be operated without a conventional mixer. All mixing and processing can be done either in the 896mk3, in the computer with audio software, or both. During recording, you can use the 896mk3's

CueMix™ FX mixer to apply reverb, EQ and compression to what you are recording and monitor it via the main outs, headphone outs, or any other output pair. You can control everything from the included CueMix Console software.

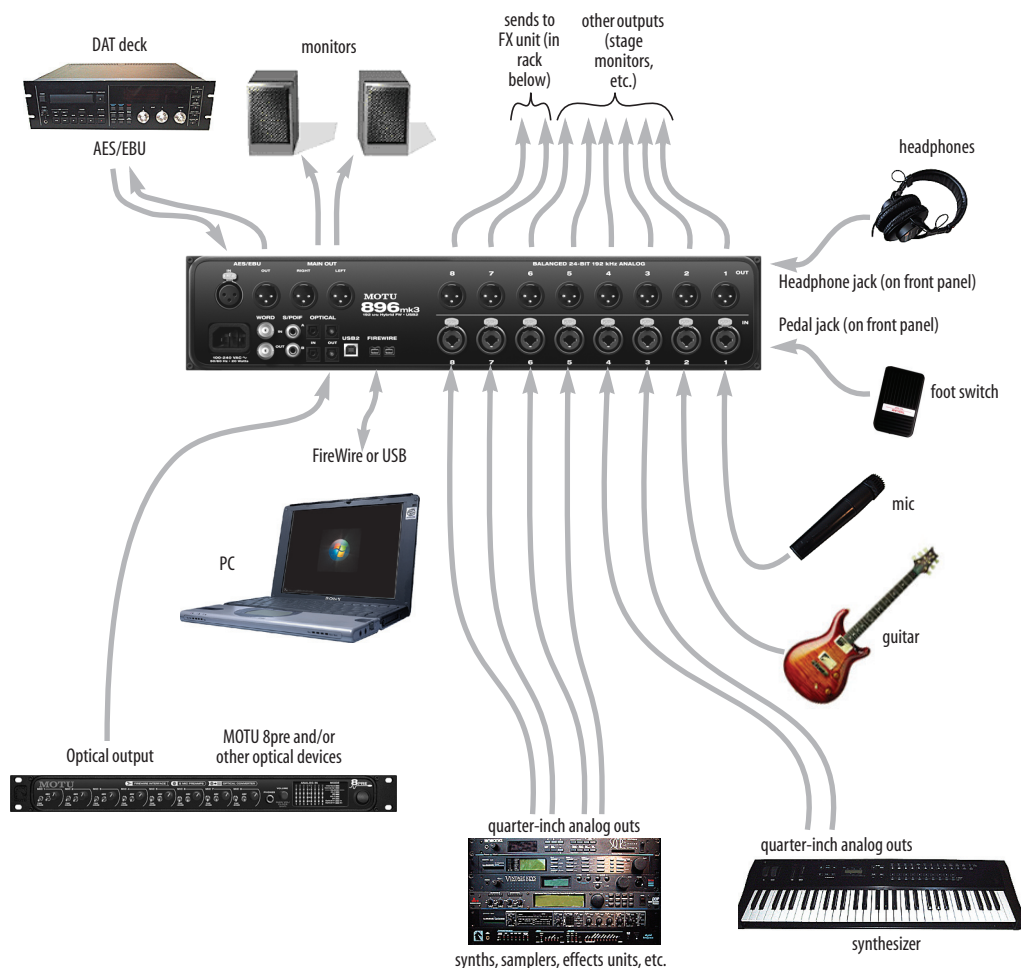


Figure 4-6: A typical 896mk3 studio setup.

OPERATING THE 896mk3 AS A CONVERTER

As explained earlier in “Using optical I/O to operate the 896mk3 as an 8-channel expander” on page 25, the 896mk3 can serve as a multi-channel analog-to-digital converter when disconnected from the computer and instead connected to another device equipped with an ADAT optical input. For example, you could connect the 896mk3 optical output to the optical input on another MOTU audio interface, such as a Traveler, 828mk3 or even another 896mk3. The 896mk3 then serves as a multi-channel expander that adds additional mic/analog and digital inputs to the interface. The

benefit of connecting the 896mk3 in this manner (instead of as another FireWire interface) is that you can seamlessly integrate the 896mk3’s inputs into the on-board no-latency CueMix monitor mixing in the interface, since the 896mk3’s inputs are fed into CueMix via the interface’s optical inputs.

If the device to which you are connecting the 896mk3 supports 2x optical sample rates (88.2 or 96 kHz), you can also use both banks of connectors as discussed in “ADAT optical operation at 88.2 or 96 kHz” on page 24.

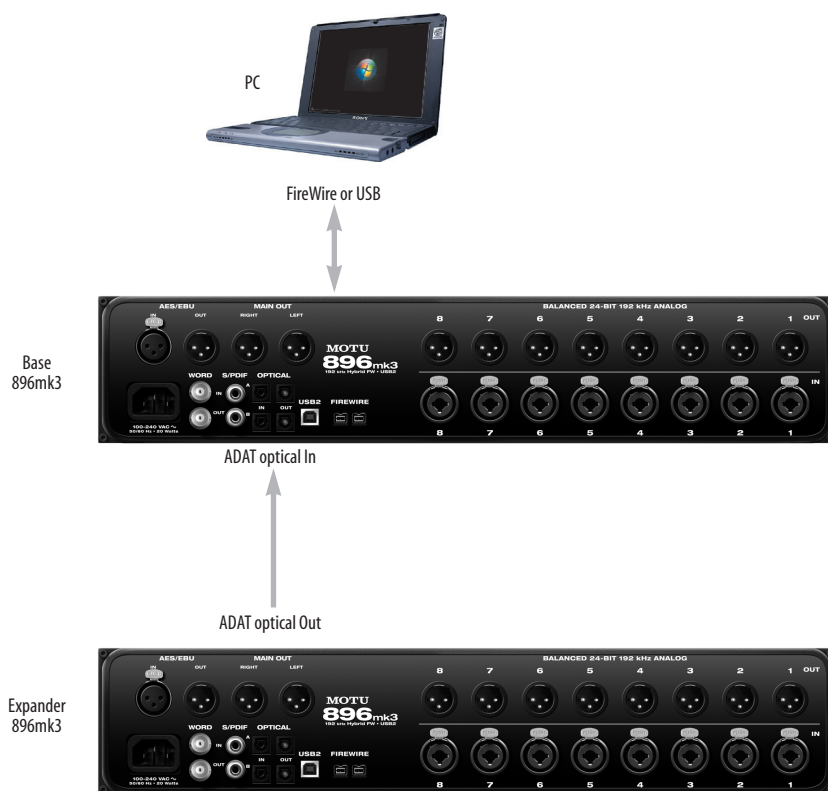


Figure 4-7: Using the 896mk3 as an optical expander. In this example, it is connected to another 896mk3.

MAKING SYNC CONNECTIONS

If you connect devices digitally to the 896mk3, or if you need to synchronize the 896mk3 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 896mk3?

If you will be using only the 896mk3's analog inputs and outputs (and none of its digital I/O), and you have no plans to synchronize your 896mk3 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 5, "MOTU Audio Console" (page 39), where you'll open MOTU Audio Console and set the *Master Clock Source* setting to *Internal* as shown below. For details, see chapter 5, "MOTU Audio Console" (page 39).

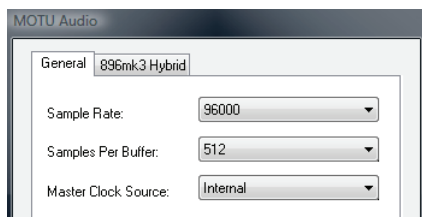


Figure 4-8: You can run the 896mk3 under its own internal clock when it has no digital audio connections and you are not synchronizing the 896mk3 system to an external time reference such as SMPTE time code.

Situations that require synchronization

There are three general cases in which you will need to resolve the 896mk3 with other devices:

- Synchronizing the 896mk3 with other digital audio devices so that their digital audio clocks are *phase-locked* (as shown in Figure 4-9)
- Resolving the 896mk3 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 896mk3'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.

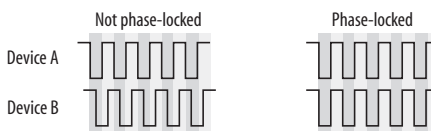


Figure 4-9: When transferring audio, two devices must have phase-locked audio clocks to prevent clicks, pops or other artifacts.

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.

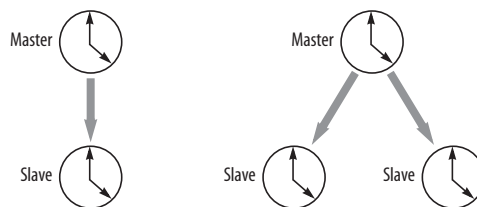


Figure 4-10: To keep the 896mk3 phased-locked with other digital audio devices connected to it, choose a clock master.

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 896mk3.

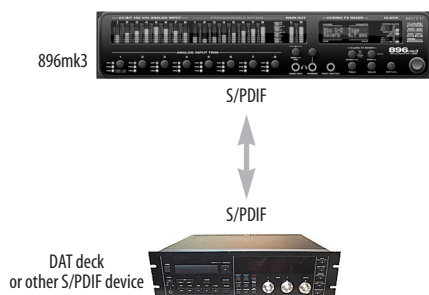
SYNCING S/PDIF DEVICES

S/PDIF devices will sync to the 896mk3 in one of two ways:

- Via the S/PDIF connection itself
- Via word clock

S/PDIF devices with no word clock

If your S/PDIF device has no word clock sync connectors, just connect it to the 896mk3 via the S/PDIF connectors. When the device records S/PDIF audio (from the 896mk3), it will simply synchronize to the clock provided by the audio input.



On the other hand, when you transfer audio from the S/PDIF device into the 896mk3, you'll have to slave the 896mk3 to its S/PDIF input. If you have other digital audio devices connected to the 896mk3, and they are not slaved directly to the 896mk3 itself, you may hear clicks and pops resulting from their unsynchronized audio clock. If so, just turn them off during the transfer.

S/PDIF devices with word clock

If your S/PDIF device has a Word Clock input, slave the S/PDIF device to the 896mk3 via their word clock connection. You can then freely transfer audio between the 896mk3 and the S/PDIF device.

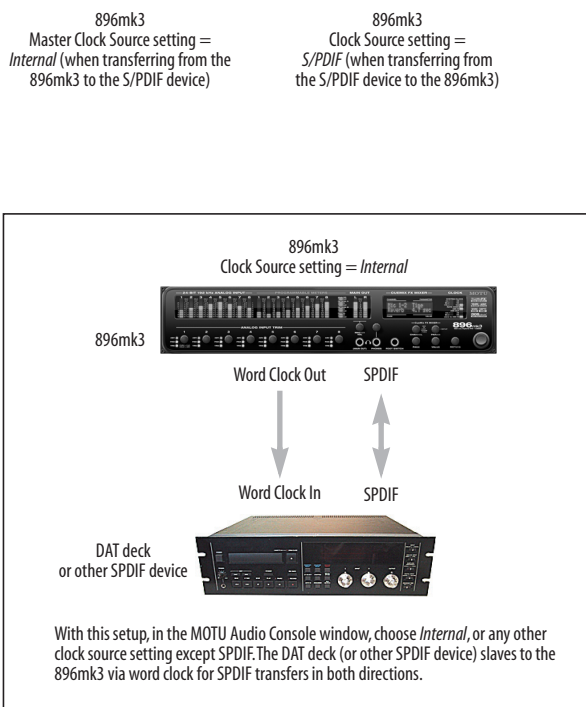


Figure 4-11: Two setups for synchronizing a S/PDIF device with the 896mk3. In the top diagram, sync is achieved via the S/PDIF connection itself. In this case, you have to choose S/PDIF as the 896mk3's clock source when recording from the S/PDIF device. If you don't want to have to worry about switching the Clock Source setting depending on the direction of the S/PDIF transfer, you can slave the S/PDIF device to word clock from the 896mk3 or vice versa (not shown). The Word Clock connection maintains sync, regardless of the direction of the transfer.

SYNCING OPTICAL DEVICES

When connecting an optical device, make sure that its digital audio clock is phase-locked (in sync with) the 896mk3, as explained in “Making sync connections” on page 28. There are two ways to do this:

1. Resolve the optical device to the 896mk3
2. Resolve the 896mk3 to the optical device

For 1), choose *Internal* (or any other clock source except *ADAT optical*) as the clock source for the 896mk3 in MOTU Audio Console.

For 2), choose either *ADAT Optical A* or *ADAT Optical B* as the 896mk3’s clock source (Figure 4-12). Be sure to choose the optical port that the device is connected to.

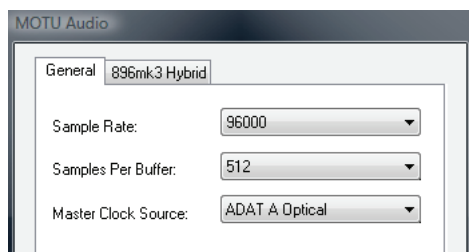


Figure 4-12: Resolving the 896mk3 to an optical device.

For details about using the clock source setting and the MOTU Audio Console software in general, see chapter 5, “MOTU Audio Console” (page 39).

Using word clock to resolve optical devices

If the optical device you are connecting to the 896mk3 has word clock connectors on it, you can use them to resolve the device to the 896mk3, similar to the diagram shown in Figure 4-11 on page 29 for S/PDIF devices with word clock. Also see “Syncing word clock devices” on page 33.

SYNCING AES/EBU DEVICES

If you would like to transfer stereo audio digitally between the 896mk3 and another device that has AES/EBU, connect it to the 896mk3’s AES/EBU jacks with balanced, AES/EBU grade audio cables.

AES/EBU clock and sample rate conversion

The 896mk3 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 896mk3 at a sample rate that is completely different than the 896mk3 system clock rate.
- Transfer digital audio into the 896mk3 without the need for any external synchronization arrangements.
- Transfer digital audio out of the 896mk3 at double or half the 896mk3 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 discussed earlier in “Be sure to choose a digital audio clock master” on page 28 and Figure 4-10 on page 28. Otherwise, you’ll hear clicks, pops, and distortion in the audio, or perhaps no audio at all. Phase lock ensures a clean digital audio transfer.

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 4-13, which shows the clocking going on when you transfer digital audio from the 896mk3 (AES/EBU OUT) to a DAT deck (AES/EBU IN) using SRC. Notice that with SRC, the DAT deck is not slaved to the 896mk3’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 896mk3 for a clean digital audio transfer (unless it has its own sample rate converter on its AES/EBU input).

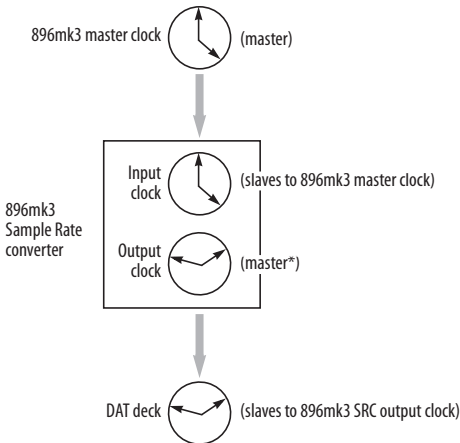


Figure 4-13: Clock relationships when sending audio from the 896mk3 to a DAT deck using sample rate conversion. The DAT deck needs to be slaving to its AES/EBU input. *Note: the 896mk3 AES/EBU output can actually be clocked from a number of different sources. In this example, it is resolved to the 896mk3 system clock. For details about other possible clock sources, see “Clocking scenarios for AES/EBU output” on page 32.

System clock, AES clock & rate convert settings

When you are setting up AES/EBU input and output with the 896mk3, pay careful attention to the following settings in MOTU Audio Console (see the quick reference overview on page 9):

- Clock source
- Sample rate conversion

These options are mentioned briefly in the following sections. For further details, see “Master Clock Source” on page 41 and “Sample Rate Convert” on page 44.

Clocking scenarios for AES/EBU input

There are three possible clocking scenarios for the 896mk3 AES/EBU input:

1. Simple transfer (slave the 896mk3 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 896mk3 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
896mk3 clock source setting	AES/EBU	Any setting except AES/EBU	Word Clock
Sample rate conversion setting	None	AES In	None
Required 896mk3 cable connections	AES/EBU In	AES/EBU In	AES/EBU In and Word Clock In
Are the devices continuously resolved?	Yes	No	Yes
Is the signal being sample rate converted?	No	Yes	No
Example application	Simple digital transfer into the 896mk3 from DAT deck or digital mixer.	Transfer from digital mixer running at a different sample rate.	Both the 896mk3 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)

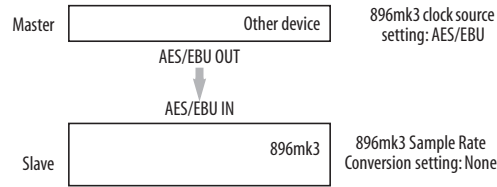


Figure 4-14: Slaving the 896mk3 to an AES/EBU device. For the 896mk3's clock source, choose 'AES/EBU'.

AES/EBU input with rate conversion

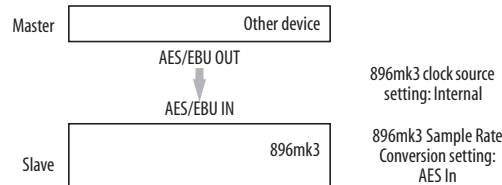


Figure 4-15: Rate-converting AES/EBU input.

AES/EBU input with word clock

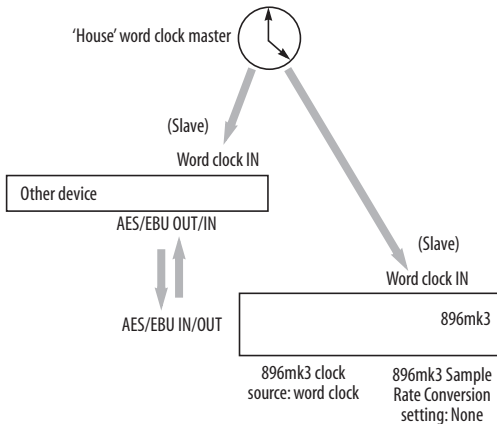


Figure 4-16: In this scenario, the 896mk3 and other AES/EBU device are both resolved to one another via a third master word clock source.

Clocking scenarios for AES/EBU output

The 896mk3 AES/EBU output can also employ sample rate conversion. The output options, shown below in Figure 4-17, are briefly summarized in the following sections. For further details, see "Sample Rate Convert" on page 44.

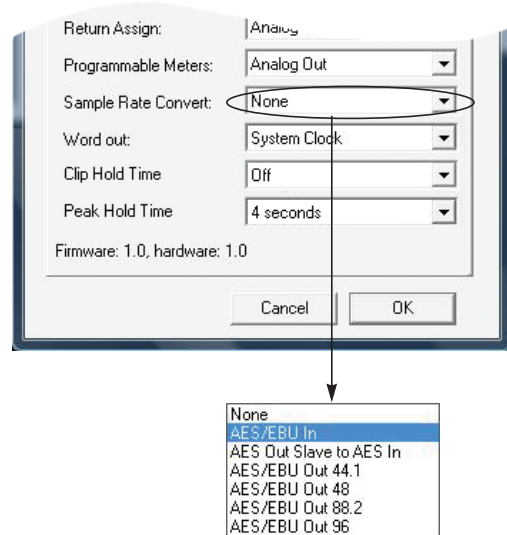


Figure 4-17: The Sample Rate Conversion option in MOTU Audio Console gives you access to the AES/EBU output clock options.

None

To make the AES/EBU output sample rate match the System sample rate, choose *None* (Figure 4-17). 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 896mk3's AES/EBU input, choose *AES Out Slave to AES In* (Figure 4-17). This setting requires a connection to the 896mk3's AES/EBU input from a device that is transmitting an AES/EBU clock signal.

Be careful when both the 896mk3's AES/EBU input and output are connected to the same external device: this option is likely to create a clock loop.

AES/EBU Out options

Choose the desired *AES/EBU Out* sample rate options (Figure 4-17) when the desired AES/EBU output rate needs to be completely different than the system clock rate or the AES In clock rate.

SYNCING WORD CLOCK DEVICES

The 896mk3 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 28). 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 4-18 and Figure 4-19.



Figure 4-18: Slaving another digital audio device to the 896mk3 via word clock. For the 896mk3 clock source, choose any source besides word clock, as it is not advisable to chain word clock.

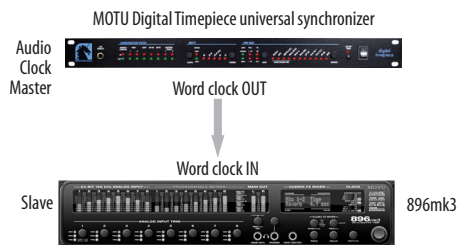


Figure 4-19: Slaving the 896mk3 to word clock. For the 896mk3 clock source, choose 'Word Clock In'.

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

All MOTU FireWire audio interfaces that support 96 kHz operation have the ability to slave to a word clock signal running at either one half or one quarter of their current clock rate. For example, the 896mk3 could be running at 96 kHz while slaving to a 48 kHz word clock signal. Similarly, the 896mk3 could run at 88.2 kHz and slave to 44.1 kHz word clock.

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

- the same as the 896mk3 clock
- half of the 896mk3 clock

Forcing a 1x word out rate

The 896mk3 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 896mk3 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 45.

SYNCING TO SMPTE TIME CODE

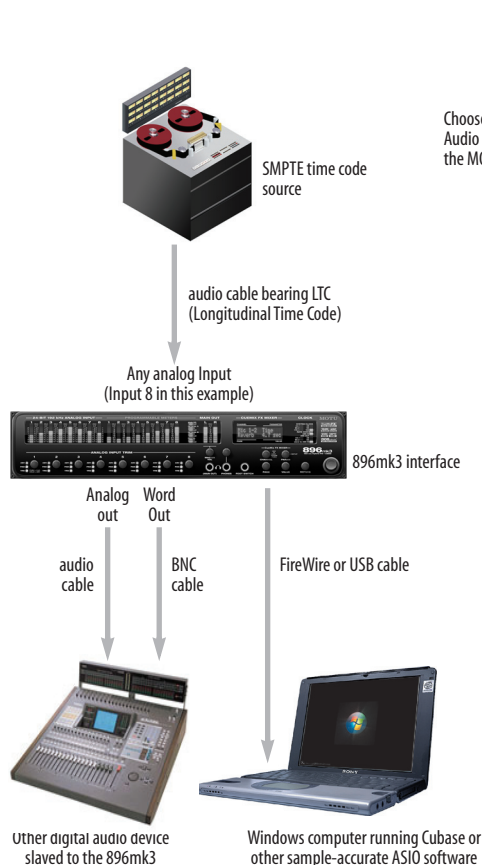
The 896mk3 system can resolve directly to SMPTE time code. It can also generate time code and word clock, under its own clock or while slaving to time code. Therefore, the 896mk3 can act both as an audio interface and as a digital audio synchronizer to which you can slave other digital audio devices. You can use the 896mk3 to slave your audio software to SMPTE as well, via sample-accurate sync (if your host software supports it).

Use this setup if you have:

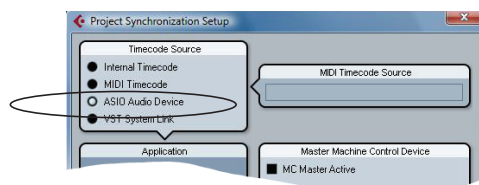
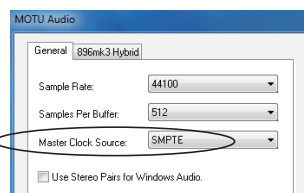
- ✓ A SMPTE time code source, such as a multitrack tape deck.
- ✓ An 896mk3 by itself, OR with another slaved device (such as a digital mixer).
- ✓ Host software that supports sample-accurate sync (such as Cubase or Nuendo).

This setup provides:

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



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



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



Launch the MOTU SMPTE Console and specify the *SMPTE Source*, which is the input receiving the SMPTE time code. Also, confirm that the *Clock Source/Address* is *SMPTE/SMPTE*. For details about the other settings, see chapter 10, "MOTU SMPTE Console" (page 119).

Figure 4-20: Connections for synchronizing the 896mk3 directly to SMPTE time code.

CONNECTING MULTIPLE MOTU FIREWIRE INTERFACES

You can daisy-chain up to three 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:

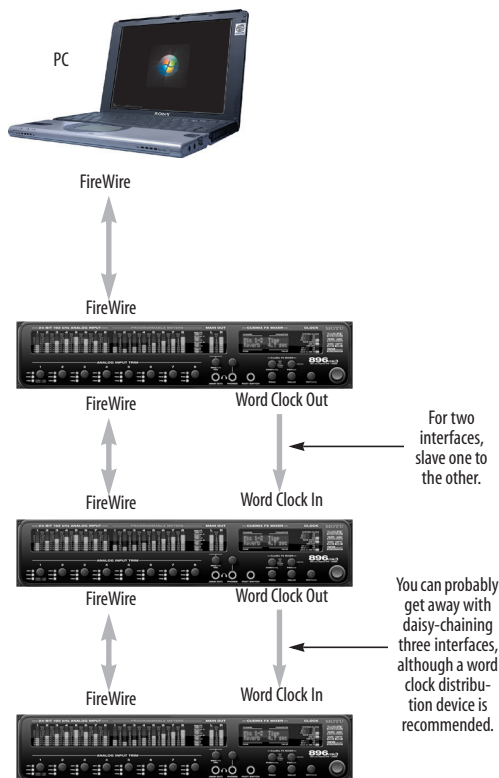


Figure 4-21: Connecting multiple 896mk3's (or other MOTU FireWire audio interfaces) to a computer.

Multiple interfaces cannot be bus-powered

Do not run the Traveler, UltraLite or other bus-powered interfaces under bus power when connecting them with other devices on the same FireWire bus.

Multiple interfaces in MOTU Audio Console

MOTU Audio Console displays the settings for one interface at a time. To view the settings for an interface, click its tab as shown below in Figure 4-22.

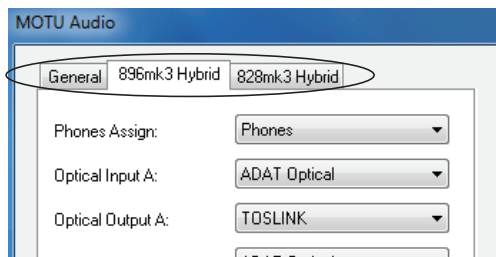


Figure 4-22: To view the settings for an interface, click its tab.

Synchronizing multiple interfaces

When multiple MOTU interfaces are connected, choose a *Master Clock Source* from the menu in the General tab (Figure 4-23). You can choose any available sync source from any connected interface (as shown in Figure 4-23), giving you a great deal of flexibility in choosing a clock source.

A separate *Clock* menu is provided for any interfaces not chosen as the master clock source, as demonstrated with the 896mk3 Hybrid in Figure 4-23. You can either resolve the interface to the *Master Clock Source* or to one of its own digital inputs or other sync sources. The latter scenario is useful for externally resolving the interface with the the interface chosen as the Master Clock Source.

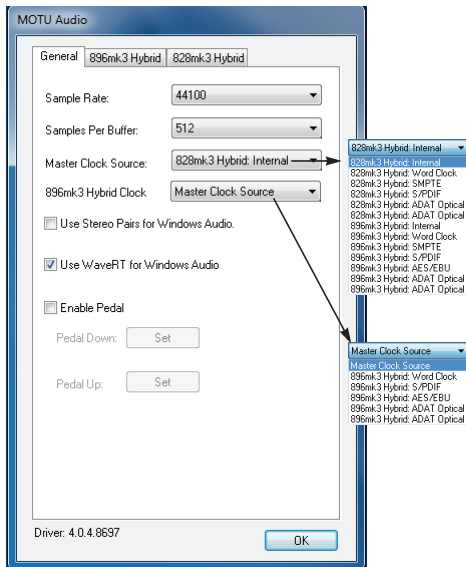


Figure 4-23: Choosing clock sources with multiple interfaces.

Connecting other MOTU FireWire interfaces

You can mix and match multiple 896mk3 Hybrid interfaces with other MOTU FireWire interfaces using a standard FireWire hub. You can daisy-chain 828mkIIs, 828mk3s, 896HDs, 896mk3s, Travelers, and Traveler-mk3s, which all have two FireWire ports convenient for daisy-chaining. Up to three interfaces can be combined on one FireWire bus.

Operating multiple FireWire interfaces at high sample rates

Three MOTU FireWire interfaces can operate at 44.1 or 48kHz on a single FireWire bus, although you may have to disable optical banks to conserve FireWire bus bandwidth. At the 2x samples rates (88.2 or 96kHz) and 4x sample rates (176.4 and 192kHz), you can operate no more than two 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”) adapter 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 their performance with your host computer.

Part 2

Using the

896mk3 Hybrid

CHAPTER 5 MOTU Audio Console

OVERVIEW

MOTU Audio Console gives you access to basic 896mk3 hardware settings, such as sample rate, clock source, optical format and more.

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ACCESSING THE 896MK3 SETTINGS

There are several ways to access 896mk3 settings:

- Open MOTU Audio Console from the Windows Start menu > Programs > MOTU.
- From within your host audio software's audio driver preferences. For example, in Cubase or Nuendo, open the Device Setup window, click *VST Audio System* and choose *MOTU Audio ASIO* from the *ASIO Driver* menu. Then click the *MOTU Audio ASIO* item in the list and click the *Control Panel* button.
- ☛ Most applications provide access only to basic settings such as sample rate and clock source. For access to all settings, open MOTU Audio Console using one of the techniques above.
- From the front panel LCD as explained in chapter 6, "Front Panel Operation" (page 47).

896mk3 tab settings

The 896mk3 tab (Figure 5-1) provides settings that apply to a specific 896mk3 interface. If you have several 896mk3 (or other MOTU) interfaces connected, you'll see a separate tab for each one.

General tab settings

The General tab provides settings that apply to all connected MOTU FireWire and USB Audio interfaces.

'GENERAL' TAB SETTINGS

Sample Rate

Choose the desired *Sample Rate* for recording and playback. The 896mk3 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 896mk3 match the 896mk3's sample rate. Also make sure that your digital audio synchronizer matches it as well. At the 4x sample rates (176.4 or 192kHz), all digital I/O on the 896mk3 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 MOTU Audio Console.

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

At the 4x sample rates (176.4 or 192kHz), operation of the 896mk3 is restricted, due to the higher audio bandwidth demands, as follows:

- All digital I/O is disabled (there is no optical, S/PDIF or AES/EBU input/output).
- The 896mk3 provides 8 channels of analog input and 8 channels of analog output, simultaneously.
- The *stereo return* bus, as described in “Return Assign” on page 44, can only be assigned to one of the four available analog output pairs or main outs.
- The PHONES jack can only be assigned to one of the four available analog output pairs or the main outs.
- The MAIN OUT jack can only be assigned to one of four analog output pairs or the main outs.

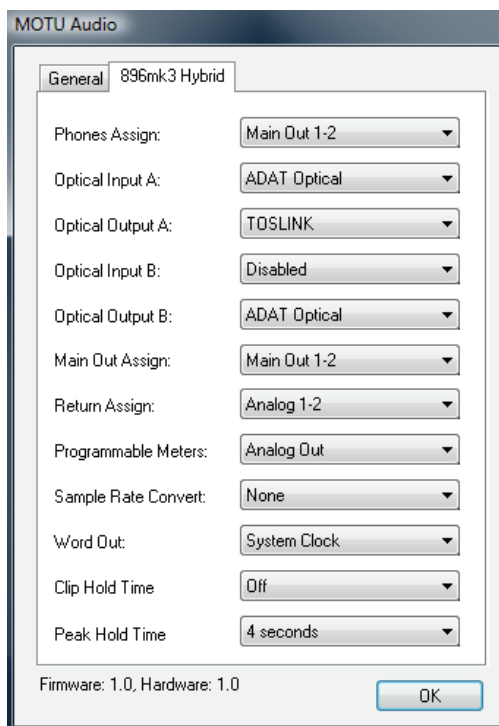
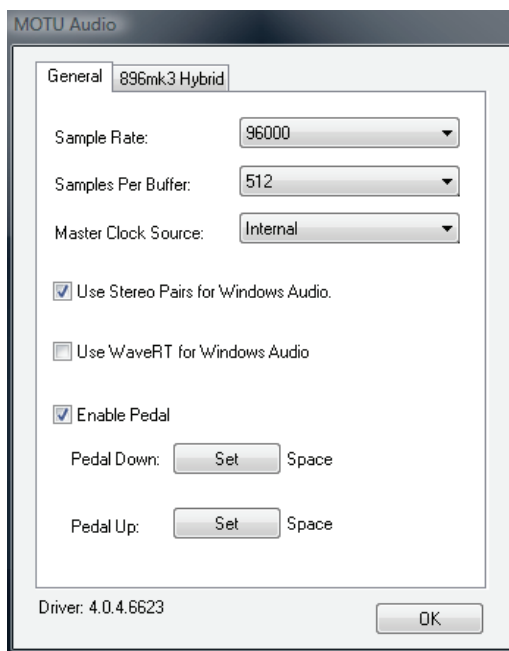


Figure 5-1: MOTU Audio Console gives you access to all of the settings in the 896mk3 hardware.

Master Clock Source

The *Master Clock Source* determines the digital audio clock that the 896mk3 will use as its time base. For a complete explanation of synchronization issues, see “Making sync connections” on page 28. The following sections briefly discuss each clock source setting.

Internal

Use the *Internal* setting when you want the 896mk3 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 896mk3 system on its internal clock, and then slave the DAT deck to the 896mk3 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 896mk3’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 “Making sync connections” on page 28.

Word Clock In

The *Word Clock In* setting refers to the Word Clock In BNC connector on the 896mk3 rear panel. Choosing this setting allows the 896mk3 to slave to an external word clock source, such as the word clock output from a digital mixer.

S/PDIF

The *S/PDIF* clock source setting refers to the S/PDIF RCA input jack on the 896mk3. This setting allows the 896mk3 to slave to another S/PDIF device.

Use this setting whenever you are recording input from a DAT deck or other S/PDIF device into the 896mk3. It is not necessary in the opposite direction (when you are transferring from the 896mk3 to the DAT machine).

For further details about this setting, see “Syncing S/PDIF devices” on page 29.

AES/EBU

The *AES/EBU* clock source setting refers to the AES/EBU input connector on the 896mk3. This setting allows the MOTU 896mk3 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 896mk3. It is not necessary in the opposite direction (when you are transferring from the 896mk3 to the DAT machine).

For further details about this setting, see “Syncing AES/EBU devices” on page 30.

ADAT optical A / B

The *ADAT optical* clock source settings (*ADAT Optical A* and *ADAT Optical B*) refer to the clock provided by the 896mk3’s two optical inputs, when either one is connected to another optical device. These two settings only appear in the Master Clock Source menu when their corresponding optical bank input is enabled and set to the *ADAT Optical* format, as explained in “Optical input/output” on page 44.

This setting can be used to resolve the 896mk3 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 optical device that has no way of synchronizing digitally to the 896mk3 or an external

synchronizer. In this case, the *ADAT Optical* clock source setting lets you slave the 896mk3 to the device itself via its digital input to the 896mk3.

This setting is also useful if you just need to make a simple, click-free digital transfer between the 896mk3 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 30.

TOSLINK A / B

The *TOSLINK* clock source settings (*TOSLINK A* and *TOSLINK B*) refer to the clock provided by the 896mk3’s two optical inputs, when either one is connected to another optical device. These two settings only appear in the Master Clock Source menu when their corresponding optical bank input is enabled and set to the *TOSLINK* format, as explained in “Optical input/output” on page 44.

The *TOSLINK* clock source setting refers to the clock provided by an optical S/PDIF device connected to the 896mk3’s optical input. This setting can be used to slave the 896mk3 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 a TOSLINK-compatible device that has no way of synchronizing digitally to the 896mk3 or an external synchronizer such as the Digital Timepiece. In this case, the *TOSLINK* clock source setting lets you slave the 896mk3 to the other device via the 896mk3’s optical input.

This setting is also useful if you just need to make a simple, click-free digital transfer between the 896mk3 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, see “Syncing optical devices” on page 30.

SMPTE

Choose this setting to resolve the 896mk3 directly to SMPTE time code (LTC) being received via the 896mk3’s quarter-inch SMPTE input jack. For details, see “Syncing to SMPTE time code” on page 34 and chapter 10, “MOTU SMPTE Console” (page 119).

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 guitar input that you would like to run through an amp simulation plug-in that you are running in your host audio software. When doing so, you may hear or feel some “sponginess” (delay) between the source and the processed signal. If so, don’t worry. This effect only affects what you hear: it is not present in what is actually recorded.

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

☞ If you don’t need to process an incoming live signal with software plug-ins, you can monitor the signal with no delay at all using CueMix FX, which routes the signal directly to your speakers via hardware. For details, see chapter 9, “CueMix FX” (page 71).

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

- The strain on your computer’s CPU
- The delay you hear when routing a live signal through your host audio software plug-ins

- How responsive the transport controls are in your software

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

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

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

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

Monitoring live inputs without plug-in effects

As mentioned earlier, CueMix FX allows you to monitor dry, unprocessed live inputs with no delay at all. For complete details, see chapter 8, "Reducing Monitoring Latency" (page 65).

Use stereo pairs for Windows audio

Some legacy Windows audio applications require audio channels to be grouped in stereo pairs. If you are using old Windows audio software, and you are having issues accessing Audio Express channels from within the software, try enabling the *Use Stereo Pairs for Windows Audio* option.

Use WaveRT for Windows audio

If your host software supports WaveRT, enable this option. Consult your host application documentation for details.

Enable Pedal

Check the Enable Pedal option if a foot switch is connected to the 896mk3 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.)

896MK3 TAB SETTINGS

Phones Assign

The *Phones Assign* setting lets you choose what you will hear from the headphone jack. Choose *Main Out 1-2* if you'd like to route the Main Out stereo pair to the PHONES jack. Choose *Phones 1-2* 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 FX mix busses.

Optical input/output

The *Optical input* and *Optical output* settings let you choose between ADAT optical ('lightpipe') and S/PDIF optical ('TOSLINK') as the format for the 896mk3's two banks of optical input and output (Bank A and Bank B). Choose the format that matches the device connected. If you are not using the optical connections, it is recommended that you turn them off (by choosing *Disabled* from the menu) to reduce bandwidth and processing overhead. Note that you can operate each optical port independently. For example, you could use the ADAT optical format on the Bank A input (for a digital mixer) and optical S/PDIF on the Bank A output (for a DAT deck).

Main Out Assign

Choose *Main Outs* from the *Main Out Assign* menu to treat the Main Outs as their own independent output pair. Choose any other output pair to cause the MAIN OUT jacks to mirror (duplicate) the output pair you choose.

Return Assign

The *Return Assign* menu lets you choose any pair of 896mk3 audio outputs. The audio signal from this output pair is then sent back to the computer via the *Stereo Return 1-2* bus. This stereo return bus from the 896mk3 appears in your host software alongside all other 896mk3 inputs, wherever your host software lists them.

The 896mk3 stereo return bus can be used for a variety of purposes. For example, you could use it to send a final mix being played through the 896mk3 back to the computer, where you could record it for mastering or archiving purposes.

As another example, you could use the stereo return bus to capture tracks played from your host software, along with live inputs being routed directly through the 896mk3 hardware via CueMix FX (with or without CueMix effects processing on the live inputs).

Programmable Meters

This option lets you choose which bank you wish to monitor with the eight programmable meters on the MOTU 896mk3 front panel. Your choices are: Analog Out, Optical A In, Optical A Out, Optical B In, Optical B Out, or Digital In/Out. This last option displays AES/EBU input, S/PDIF input, AES/EBU output and S/PDIF output, in order from left to right. You can also adjust this setting by turning the METERS knob on the 896mk3 front panel.

Sample Rate Convert

The Sample Rate Convert option (Figure 4-17 on page 32) lets you control AES/EBU sample rate conversion. Sample rate conversion is available when the 896mk3 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 samples rates (176.4 and 192kHz). Each option is explained below.

None

To make the AES/EBU output sample rate match the System sample rate, choose *None* (Figure 4-17 on page 32). 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 896mk3's AES/EBU input, choose *AES Out Slave to AES In* (Figure 4-17 on page 32). This setting requires a connection to the 896mk3's AES/EBU input from a device that is transmitting an AES/EBU clock signal.

⚠ Be careful when both the 896mk3's AES/EBU input and output are connected to the same external device: this option is likely to create a clock loop.

AES/EBU Out options

Choose the desired *AES/EBU Out* sample rate options (Figure 4-17 on page 32) when the desired AES/EBU output rate needs to be completely different than the system clock rate or the AES In clock rate. For further details about this option, see “Sample Rate Convert” on page 44.

Clip Hold Time

The *Clip Hold Time* option controls how long the top red LED on the front panel metering remains illuminated after clipping occurs.

Peak Hold Time

The 896mk3 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.

Word Out

The *Word Out* menu appears when the 896mk3 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 896mk3 were operating at 176.4kHz, choosing the *Force 44.1/48kHz* option would produce word clock output at 44.1kHz.

CHAPTER 6 Front Panel Operation

OVERVIEW

The 896mk3 Hybrid offers complete front-panel programming via six rotary encoders and a 2x16 backlit LCD display. All 896mk3 settings can be accessed via these front-panel controls.

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ANALOG INPUT CONTROLS

The 896mk3 front panel analog input controls (Figure 6-1) allow you to work with a wide variety of recording situations.

For information about connections and settings, see “Analog inputs” on page 23 in the installation chapter.

For information about the many settings available for the mic/guitar inputs, see:

- “The Inputs tab” on page 77
- “The channel settings section” on page 81
- “The Channel tab” on page 82
- “The EQ tab” on page 84
- “The Dynamics tab” on page 91

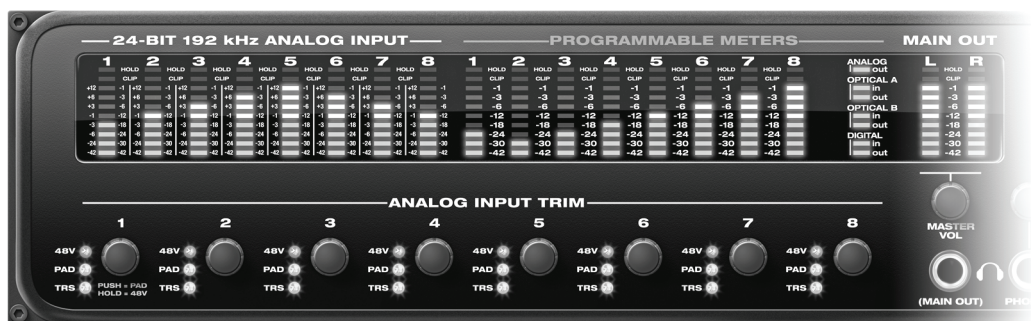


Figure 6-1: The 896mk3 Hybrid front panel mic/instrument input controls.

MASTER VOL AND “(MAIN OUT)” PHONE JACK

The phone jack labeled (MAIN OUT) (Figure 6-1) is a standard quarter-inch stereo headphone jack. Its output is hard-wired to mirror the XLR main outs on the rear panel. From the factory, the MASTER VOL knob above it controls the main outs and this jack, but MASTER VOL can be programmed to control any combination of outputs. See “The Monitor Group” on page 95 for details. Push the knob once to view the current volume setting in the LCD display; push it again to mute the monitor group; push a third time to return to the previous volume.

☞ Note that if the Monitor Group is programmed to not include the main outs, the MASTER VOL knob will no longer control the volume of the phone jack, either.



Figure 6-2: the phone jacks.

PHONES

From the factory, the PHONES jack (Figure 6-2) is a discrete output at 44.1/48 kHz, but it can mirror any other output pair (digital or analog) or serve as its own independent output. For example, at 88.2/96 kHz, it defaults to mirroring the XLR main outs. At 176.4/192 kHz, it defaults to mirroring analog outputs 1-2.

As the primary phone jack, it has its own dedicated volume knob.

The LCD provides detailed feedback as you turn the knob (Figure 6-7). To view the current setting without changing it, just push the knob (without turning it).

If you would like the (MAIN OUT) phone jack and the PHONES jack to output the same signal, assign the PHONES output to mirror the main outs. In this configuration, both the MASTER VOL and the PHONES volume knobs control the PHONES volume.

ANALOG INPUT METERS WITH V-LIMIT™ COMPRESSOR

All eight inputs are equipped with V-Limit™, a hardware limiter. With the limiter turned off, signals that hit zero or above will clip (a hard digital clip). However, with V-Limit enabled, signals can go as high as +12 dB above zero with no digital clipping. If the signal then goes above +12 dB, it will clip, even with V-Limit engaged. In either case, the clip LED (above +12) will illuminate.

Two input level scales

To accommodate these two scenarios, the eight analog input meters (Figure 6-3) are labeled with two different input level scales: from -42 to -1 (when V-Limit is disabled) and -42 to +12 dB (when V-Limit is engaged).

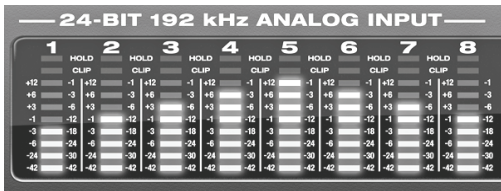


Figure 6-3: Analog input metering.

For further information about V-Limit, as well as another feature called *Soft Clip*, see “Overload protection” on page 82.

CLIP and HOLD LEDs

The CLIP and HOLD LEDs indicate when clipping has occurred (either with or without V-Limit). You can set the amount of time for the HOLD LED. See “Clip Hold Time” on page 45.

PROGRAMMABLE METERS

The programmable meter bank (Figure 6-4) lets you view the levels of any input or output bank (except analog input, which has its own bank): Analog Out, Optical A In, Optical A Out, Optical B In, Optical B Out, or Digital In/Out. This last option displays AES/EBU input, S/PDIF input, AES/EBU output and S/PDIF output, in order from left to right, as shown below.

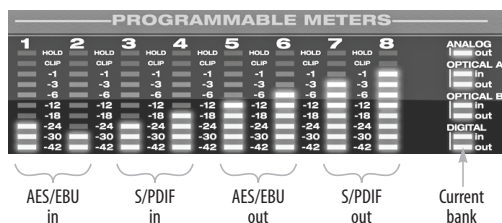


Figure 6-4: The programmable meter bank. The “Digital I/O” bank shows the AES/EBU and S/PDIF input and output as indicated above.

Choose the desired bank by turning the METERS knob on the 896mk3 front panel (Figure 6-5). The current bank is indicated by the LEDs to the right of the programmable meter bank (Figure 6-4). You can also push the METERS knob to view the currently selected bank in the LCD.

You can also specify the bank in MOTU Audio Console. See “Programmable Meters” on page 44.



Figure 6-5: The Meters knob.

PUSH-BUTTON ROTARY ENCODERS

All of the knobs shown in Figure 6-6 are push-button digital rotary encoders. In many cases, you can either push the knob or turn it to make a setting or toggle the LCD display (depending on the encoder and setting).

MULTI-FUNCTION LCD DISPLAY

The multi-function LCD (Figure 6-6) provides access to the many 896mk3 settings, as well as visual feedback of the current parameter being modified.

Parameter “zooming”

For many settings, the LCD temporarily “zooms in” to display a long-throw meter and alpha-numeric display to give you precise, real-time feedback as you adjust the setting. For example, if you change the headphone volume, the LCD will display a level meter and gain reduction reading that updates as you turn the volume knob (Figure 6-7). After a brief time-out, the display returns to its previous state before you turned the volume knob.

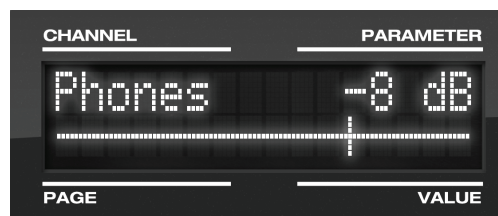


Figure 6-7: The LCD provides feedback as you adjust volume.

Three global menus

Push the PARAM knob (Figure 6-6) to cycle the LCD among three global menus, described in the rest of this chapter:

- 896mk3 SETUP
- AUDIO (settings)
- CUEMIX (mixer)

896MK3 SETUP MENU

To access the 896mk3 SETUP menu, push the PARAM knob until you see *896mk3 SETUP* displayed in the LCD. This menu provides basic features for managing the 896mk3 hardware. Turn the PARAM knob to access each setting, explained briefly below.

LCD Contrast

Turn the VALUE knob to adjust the LCD contrast.



Figure 6-8: In setup mode, the LCD displays a setup parameter in the top row of the LCD and the current setting in the bottom row.

Save/Name Preset

An 896mk3 preset holds all current CueMix FX mix settings (everything in the CUEMIX menu). SETUP menu and AUDIO menu parameters are not included. Turn the VALUE knob to move from character to character in the preset name. Turn PAGE to change the currently flashing letter. Once you've named the preset, push VALUE to save it, turn it again to choose the desired preset slot you wish to save it to (1-16), and then push again to confirm the save. Cancel the save operation at any time by turning the PARAM knob.

Load Preset

Turn VALUE to choose the preset you wish to load (1-16) and push VALUE to load it.

AutoSave Status

All of the parameters in the 896mk3 are periodically saved automatically in its memory, so that if you power down the unit, it will power back

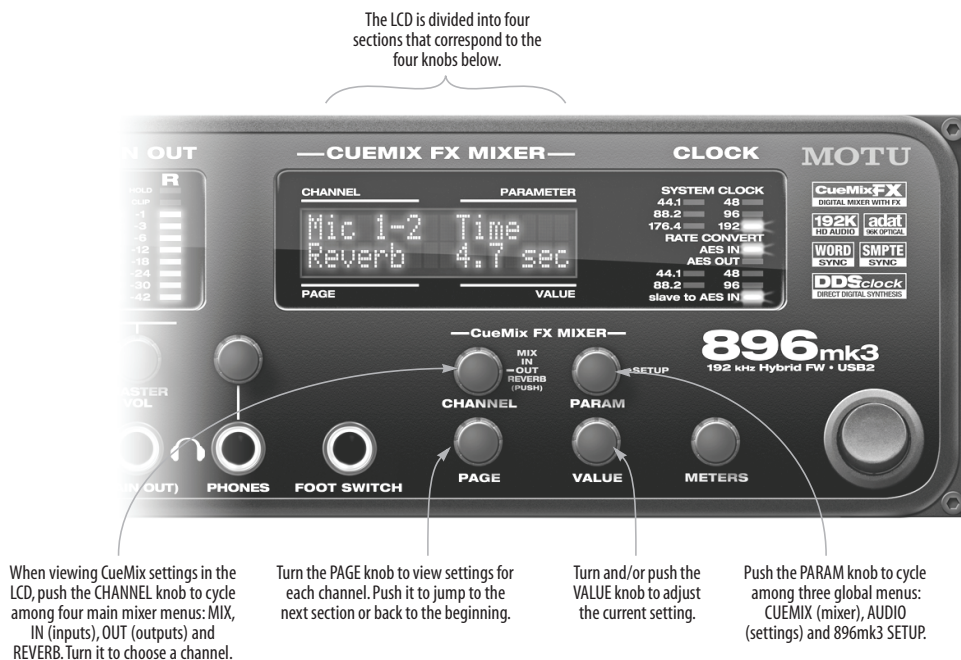


Figure 6-6: The 896mk3 Hybrid front panel controls.

up in the same state. AutoSave Status tells you the current state of this process: *Change Detected*, *Saving...* or *Saved*. If it says *Saved*, it is ready to be powered down. If it says *Change Detected* or *Saving...*, wait until it changes to *Saved*.

Interface mode

Turn VALUE to choose between FireWire or USB operation. Normally, the 896mk3 detects how it is connected to the host computer, but this setting lets you manually specify which bus to use.

Factory Defaults

Push VALUE to restore the 896mk3 hardware to its factory default settings. Push VALUE again to confirm, or turn PARAM to cancel.

AUDIO MENU

To access the AUDIO menu, push the PARAM knob until you see *AUDIO* displayed in the LCD. This menu provides basic settings such as sample rate, clock source, optical bank format (ADAT versus TOSLINK), and so on. These settings have corresponding settings in the MOTU Audio Console software, as shown below:

Audio menu item	Where to find more information
Clock Source*	"Master Clock Source" on page 41
Sample rate*	"Sample Rate" on page 40
Phones Assign	"Phones Assign" on page 43
Main Out Assign	"Main Out Assign" on page 44
Return Assign	"Return Assign" on page 44
Rate Convert	"Sample Rate Convert" on page 44
Optical In/Out A/B*	"Optical input/output" on page 44
ADAT SMUX Type	See below. Only available when the 896mk3 is operating at 88.2 or 96 kHz.
Word Clock Out	"Word Out" on page 45. Only available when the 896mk3 is operating at 88.2 or 96 kHz.
Clip Hold Time	"Clip Hold Time" on page 45
Peak Hold Time	"Peak Hold Time" on page 45

** If the 896mk3 is currently connected to a computer, this setting cannot be changed from the front-panel LCD. It must be changed in MOTU Audio Console instead. Or, you can disconnect the 896mk3 from the computer to change the Clock Source from the front panel.*

ADAT SMUX Type

When the 896mk3 is operating at either 88.2 or 96 kHz, the AUDIO menu includes an item called *ADAT SMUX Type*.

This setting lets you configure the optical ports for ADAT format operation at the 2x sample rates only (88.2 or 96 kHz).

There are two choices:

- Type I — for 2x optical connection to 3rd-party SMUX-compatible hardware products
- Type II — for 2x optical connection to MOTU products that are equipped with optical ports and support 2x operation

Turn the VALUE knob to select an optical port and push the knob to toggle between the Type I and Type II setting.

CUEMIX MENU

To access the CUEMIX menu, push the PARAM knob until you see *CUEMIX* displayed in the LCD. This menu displays the settings for the 896mk3 CueMix FX mixer.

CueMix mixer basics

It is much easier to navigate the CUEMIX menu in the LCD if you have a general understanding of the CueMix FX mixer. We strongly recommend that you review chapter 9, "CueMix FX" (page 71) before learning the LCD, especially "CueMix FX basic operation" on page 73.

CUEMIX menu organization

Mixer settings are divided into four sub-menus, which correspond to the Inputs, Mixes, Outputs and Reverb tabs in the CueMix FX software:

- IN (inputs)

- OUT (outputs)
- MIX (Mixes)
- REVERB (reverb processor)

Navigating the four main menus

To access the four main menus above, push the CHANNEL knob repeatedly. Then turn the CHANNEL knob to select the desired channel or mix.

The IN (inputs) menu

Push the CHANNEL button repeatedly until you see “I:” in the channel section of the LCD (Figure 6-9). This is similar to accessing the Inputs tab in CueMix FX console (“The Inputs tab” on page 77).

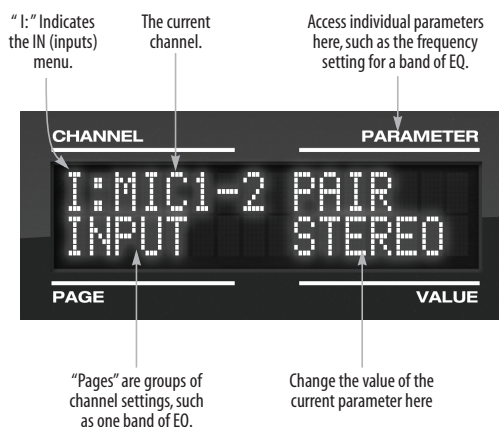


Figure 6-9: The IN (inputs) menu.

Choosing a channel

Once you see the Inputs menu (Figure 6-9) in the LCD, turn the CHANNEL knob to select the desired input that you wish to edit. This is roughly equivalent to specifying an input channel strip to work with in the Inputs tab in CueMix FX software (Figure 9-3 on page 77).

Choosing a setting to modify

Once you’ve selected an input channel, you can access the various settings for that channel using the PAGE knob and PARAMETER knobs.

Turn the PAGE knob to scroll through channel settings such as individual bands of EQ, the compressor, reverb sends, etc. Push the PAGE knob to jump to the next “section” of parameters or to jump back to the beginning of the list. This is roughly the equivalent of moving through the various channel controls in an individual input channel strip in CueMix FX (Figure 9-3 on page 77), as well as the settings in the Channel tab (Figure 9-8 on page 82).

Turn the PARAMETER knob to scroll through individual parameters, such as the frequency for the current band of EQ. This is roughly equivalent to the parameters in the EQ tab (Figure 9-10 on page 84), Dynamics tab (Figure 9-21 on page 91) and Reverb tab (Figure 9-23 on page 94) in the CueMix FX software.

Adjusting the value of a parameter

Turn the VALUE knob to adjust the value of the current PARAMETER. Some parameters have default values. If so, push the VALUE knob to cycle through them.

Inactive items

If a menu item is inactive for some reason (perhaps it doesn’t currently apply or it is disabled), it is displayed in parentheses.

Copying and pasting

EQ, dynamics and Mix Assign settings allow you to copy and paste settings between EQ bands and/or channels. Push the VALUE knob to copy, scroll to the other item and then push again to paste.

Summary of Inputs menu settings

For a summary of Inputs menu settings, see “Inputs menu” on page 55.

The OUT (Outputs) menu

Push the CHANNEL button repeatedly until you see “O:” in the channel section of the LCD (Figure 6-10). This is similar to accessing the Outputs tab in CueMix FX console (“The Outputs tab” on page 79).

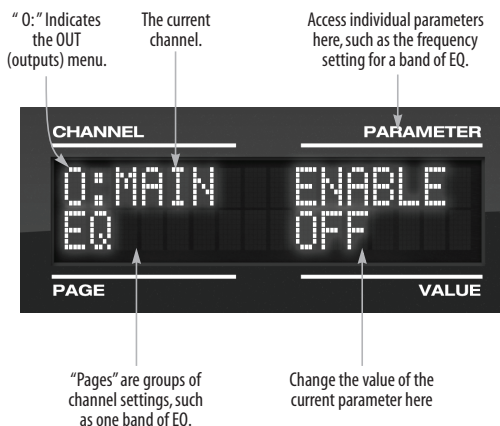


Figure 6-10: The OUT (outputs) menu.

Choosing a channel

Once you see the Outputs menu (Figure 6-10) in the LCD, turn the CHANNEL knob to select the desired output that you wish to edit. This is roughly equivalent to specifying an output channel strip to work with in the Outputs tab in CueMix FX software (Figure 9-6 on page 80).

Working with outputs in the LCD

Once you’ve selected an output channel, you can access the various settings for that channel using the PAGE knob and PARAMETER knobs. The knobs function the same as described for inputs in “Choosing a setting to modify” on page 52 and “Adjusting the value of a parameter” on page 52.

Summary of Output menu settings

For a summary of Inputs menu settings, see “Outputs menu” on page 55.

The MIX (Mixes) menu

Push the CHANNEL button repeatedly until you see “MIX 1” (or “MIX 2”, etc.) in the channel section of the LCD (Figure 6-11). This is similar to accessing the Mixes tab in CueMix FX console (“The Mixes tab” on page 75).

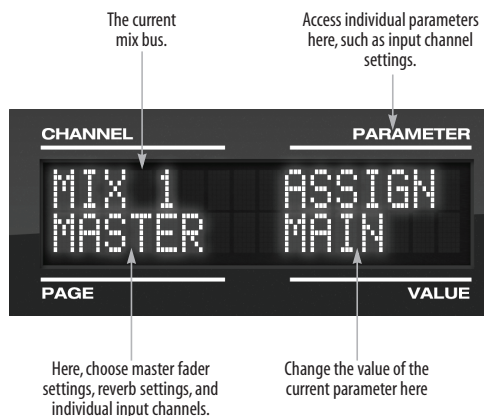


Figure 6-11: The MIX (Mixes) menu.

Choosing a mix bus

Once you see the MIX menu (Figure 6-11) in the LCD, turn the CHANNEL knob to select the desired mix that you wish to edit. This is roughly equivalent to choosing a mix in the Mixes tab in CueMix FX software (Figure 9-2 on page 75).

Working with mix busses in the LCD

Once you’ve selected a mix, you can access the various settings for that mix using the PAGE knob and PARAMETER knobs. The knobs function the same as described for inputs in “Choosing a setting to modify” on page 52 and “Adjusting the value of a parameter” on page 52. The PAGE knob scrolls through the following mix bus settings:

- MASTER — these are master fader settings, such as the master fader output assignment, master mute on/off, and the master fader volume. Access them with the PARAMETER knob.

REVERB — these are the reverb send and return controls for the bus master fader. Access them with the PARAMETER knob. Note: if the reverb processor is currently disabled, reverb is displayed in parentheses. If you are running the 896mk3 at 88.2 kHz or higher, the Reverb processor is disabled, and therefore the REVERB setting does not appear in the LCD as an option.

- Individual channels — once you scroll past MASTER and REVERB, the PAGE knob then scrolls through all available inputs for the mix bus. Once you choose an input, access its channel settings using the PARAM knob and VALUE knob.

Summary of Mix menu settings

For a summary of Mix menu settings, see “Mixes menu” on page 56.

The REVERB menu

Push the CHANNEL button repeatedly until you see “REVERB” in the channel section of the LCD (Figure 6-12). This is similar to accessing the Reverb tab in CueMix FX (“The Reverb tab” on page 94).

☞ The REVERB processor is not available at sample rates above 48 kHz. Therefore, when the 896mk3 is operating at 88.2 kHz or higher, the REVER menu does not appear in the LCD.

Working with reverb settings in the LCD

Once you’ve selected the REVERB menu, you can access all settings using the PARAMETER and VALUE knobs. The PAGE knob is not needed and is therefore disabled when editing reverb settings.

Summary of Reverb menu settings

For a summary of Reverb menu settings, see “Reverb menu” on page 56.

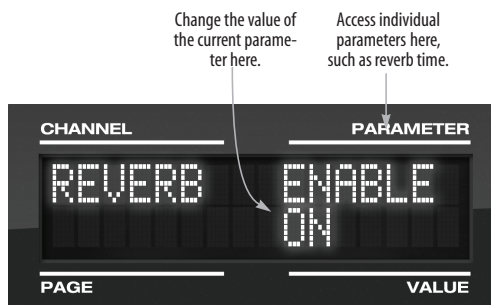


Figure 6-12: The REVERB menu.

INPUTS MENU

CHANNEL	PAGE	PARAM
INPUTS Analog 1-2 Analog 3-4 etc.	INPUT	PAIR PHASE L-R/M-S (stereo or M/S) SWAP WIDTH TRIM LIMITER LOOKAHEAD SOFTCLIP
	EQ (global)	ENABLE COPY PASTE RESET
	HPF (High-pass)	ENABLE SLOPE FREQ
	LF (Low w/shelf)	ENABLE TYPE FREQ GAIN WIDTH
	LMF (Low-mid)	ENABLE TYPE FREQ GAIN WIDTH
	MF (Mid)	ENABLE TYPE FREQ GAIN WIDTH
	HMF (High-mid)	ENABLE TYPE FREQ GAIN WIDTH
	HF (High w/shelf)	ENABLE TYPE FREQ GAIN WIDTH
	LPF (Low-pass)	ENABLE SLOPE FREQ
	DYN (Dynamics)	ENABLE COPY PASTE RESET
	COMP (Compressor)	ENABLE MODE THRESH RATIO ATTACK RELEASE TRIM
	LEVELER	ENABLE MODE REDUCE MAKEUP
	REVERB	SEND SEND PAN (mono only)

OUTPUTS MENU

CHANNEL	PAGE	PARAM
OUTPUTS Main Analog 1-2 Analog 3-4, etc.	EQ (global)	ENABLE COPY PASTE RESET
	HPF (High-pass)	ENABLE SLOPE FREQ
	LF (Low w/shelf)	ENABLE TYPE FREQ GAIN WIDTH
	LMF (Low-mid)	ENABLE TYPE FREQ GAIN WIDTH
	MF (Mid)	ENABLE TYPE FREQ GAIN WIDTH
	HMF (High-mid)	ENABLE TYPE FREQ GAIN WIDTH
	HF (High w/shelf)	ENABLE TYPE FREQ GAIN WIDTH
	LPF (Low-pass)	ENABLE SLOPE FREQ
	DYN (Dynamics)	ENABLE COPY PASTE RESET
	COMP (Compressor)	ENABLE MODE THRESH RATIO ATTACK RELEASE TRIM
	LEVELER	ENABLE MODE REDUCE MAKEUP
	REVERB	SEND RETURN
	MASTER	MONITOR TALKBACK LSNBACK

MIXES MENU

CHANNEL	PAGE	PARAM
MIXES Mix 1 Mix 2 etc.	MASTER	ASSIGN MUTE FADER COPY PASTE RESET
	REVERB	SEND RETURN
	ANALOG 1-2 (if stereo)	MUTE SOLO BAL/WID BALANCE/WIDTH FADER
	ANALOG 1 ANALOG 2 etc. (if mono)	MUTE SOLO (BAL/WID) - n/a PAN FADER
	AES 1-2	MUTE SOLO BAL/WID BALANCE/WIDTH FADER
	SPDIF	MUTE SOLO BAL/WID BAL FADER
	ADAT A1-2 ADAT A3-4 etc.	MUTE SOLO BAL/WID BALANCE/WIDTH FADER
	ADAT B1-2 ADAT B3-4 etc.	MUTE SOLO BAL/WID BALANCE/WIDTH FADER

REVERB MENU

CHANNEL	PAGE	PARAM
REVERB	---	ENABLE TIME PREDELAY WIDTH CUT Hz CUT dB ROOM REFSIZE REF LEV LO % MID % HI % LO XOVR HI XOVR SPLIT

STAND-ALONE OPERATION

All settings, including all mix settings and global settings, are saved in the 896mk3’s memory, and they remain in effect even when the 896mk3 is not connected to a computer. This allows you to use the 896mk3 as a stand-alone 8-bus mixer. You can make adjustments to any setting at any time from the front panel.

CHAPTER 7 **Configuring Host Audio Software**

OVERVIEW

The 896mk3 Hybrid provides multi-channel audio input and output for ASIO- and WDM-compatible audio applications, including Ableton Live, Avid Pro Tools, Cockos Reaper, Propellerhead Reason and Record, Steinberg Cubase and Nuendo, Cakewalk SONAR, and others.

Additionally, sample-accurate synchronization is possible with applications that support ASIO2's sample-accurate positioning protocol.

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PREPARATION

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

- chapter 3, “Installing the 896mk3 Software” (page 19)
- chapter 4, “Installing the 896mk3 Hardware” (page 21).

RUN MOTU AUDIO CONSOLE

Before you run your host audio software, launch MOTU Audio Console to configure your 896mk3 hardware.

MOTU Audio Console provides settings to configure your audio interface and enable the desired inputs and outputs. Only enabled inputs and outputs will be available to your software, so this is an important step. For complete details see chapter 5, “MOTU Audio Console” (page 39).

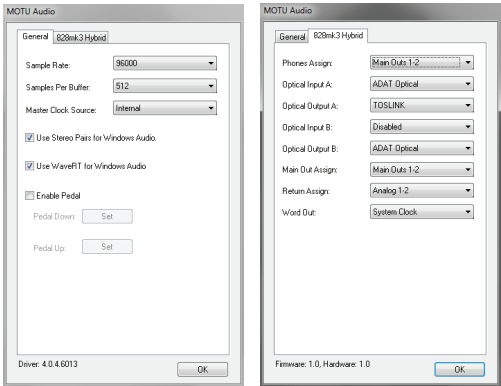


Figure 7-1: MOTU Audio Console.

For complete details about the 896mk3 settings, see chapter 5, “MOTU Audio Console” (page 39). The following sections provide a brief explanation of each 896mk3 setting.

Sample rate

Choose the desired overall sample rate for the 896mk3 system and your host audio software. Newly recorded audio will have this sample rate.

Master Clock Source

This setting is very important because it determines which audio clock the 896mk3 will follow.

If you do not have any digital audio connections to your 896mk3 (you are using the analog inputs and outputs only), and you will not be slaving your host software to external SMPTE time code, choose *Internal*.

If you have digital audio devices connected to the 896mk3, or if you are not sure about the clock source of your setup, be sure to read “Making sync connections” on page 28 and “Master Clock Source” on page 41.

If you are slaving the 896mk3 and your host software to SMPTE time code, follow the directions in “Syncing to SMPTE time code” on page 34.

☛ In order to use the 896mk3’s direct SMPTE sync (and sample-accurate sync) feature, the host audio software must support the ASIO 2.0 sample-accurate positioning protocol. Cubase and Nuendo support this protocol; for other software, check with its documentation.

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 896mk3 hardware and host audio software. For example, you might have MIDI instruments, samplers, microphones, and so on connected to the analog inputs of the 896mk3. If so, you will often be mixing their live input with audio material recorded in your audio software. See chapter 8, “Reducing Monitoring Latency” (page 65) for complete details.

Phones Assign

This 896mk3 setting lets you choose what you’ll hear from the headphone jack. For example, if you choose *Main Out 1-2*, the headphones will duplicate the main outs. Or you can choose any other output pair. If you choose *Phones 1-2*, this setting makes the headphone jack serve as its own

independent output pair. As a result, you’ll see *Phones 1-2* as an additional audio destination in your host’s audio output menus. When operating at high sample rates, the phones must mirror one of the other output pairs.

Optical input and output

To make a 896mk3 optical input or output available in your host software, choose the appropriate format (*ADAT optical* or *TOSLINK*) from the optical input and/or output menu. If you won’t be using the optical connectors, turn them off.

Main Outs Assign

Use the Main Outs Assign setting to determine what audio you will hear on the XLR main outs of the 896mk3. If you would like to treat them as their own separate output pair, choose *Main Outs*.

Return Assign

In your host audio software audio input menus, you’ll see an 896mk3 input called *Return 1-2*. This is a stereo feed from the 896mk3 that matches the signal of one of its output pairs. Use the Return Assign menu in MOTU Audio Console to choose which output pair you would like to hear on this return. This can be used, for example, to record back a final stereo mix that includes effects processing from the 896mk3 DSP (such as the *Leveler*) for reference and archiving purposes.

☛ Warning: the Return inputs can cause feedback loops! DO NOT assign this input to a track that shares the same 896mk3 output pair as the returns.

Reverb return

The 896mk3 also supplies a return to your host software that carries the output of its reverb processor. This return can be used for any purpose you wish.

CHOOSING THE MOTU AUDIO DRIVER

Once you've made the preparations described so far in this chapter, you're ready to run your audio software and enable the MOTU Audio 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 drivers that may be in your system. Choose the desired MOTU Audio driver from this menu. Several driver options are available. For a summary, see "MOTU Audio drivers" on page 20.

Pro Tools

In Avid Pro Tools, go to the Setup menu and choose *Playback Engine* as shown in Figure 7-2. Choose the *MOTU Audio ASIO* from the Current Engine menu. For information about the *H/W Buffer Size* setting, see "Adjusting the audio I/O buffer" on page 67.

🔊 Pro Tools 9 or later is required for operation with the 896mk3 Hybrid.

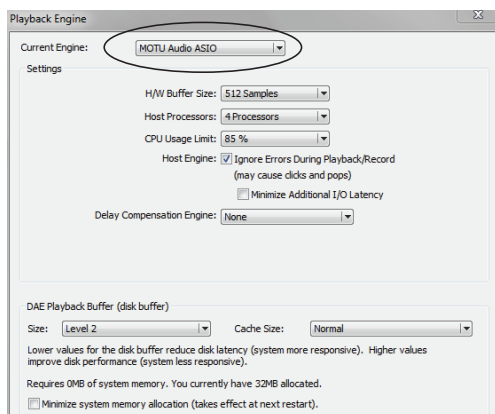


Figure 7-2: Enabling the MOTU Audio ASIO driver in Pro Tools

Cubase and Nuendo

In Cubase or Nuendo, go to the *Device Setup* window. Click the *VST Audio System* item in the Devices list and choose *MOTU Audio ASIO* from the *ASIO Driver* menu as shown below in Figure 7-3. Activate the inputs and outputs within Cubase or Nuendo as usual.

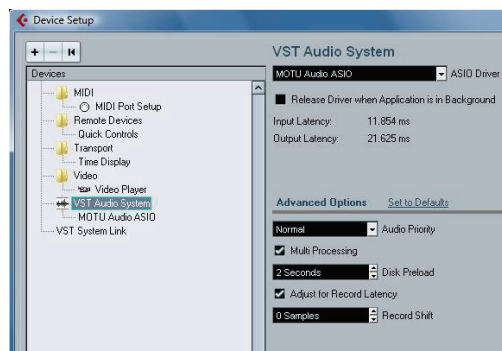


Figure 7-3: Enabling the MOTU Audio ASIO driver in Cubase

Live

In Ableton Live, access the preferences window and click the *Audio* tab. Choose *ASIO* from the *Driver Type* menu. Choose the *MOTU Audio ASIO* from the *Input Audio Device* and *Output Audio Device* menus as shown below in Figure 7-4. To enable or disable input or output channels, click the *Input Config* or *Output Config* buttons.

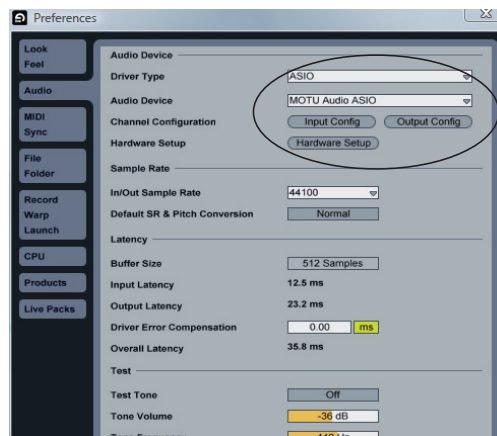


Figure 7-4: Enabling the MOTU Audio ASIO driver in Live

Reason and Record

In Propellerhead Reason or Record, go to the Preferences window, choose *Audio* preferences from the menu and choose *MOTU Audio ASIO* from the *Audio Card Driver* menu as shown below in Figure 7-5.

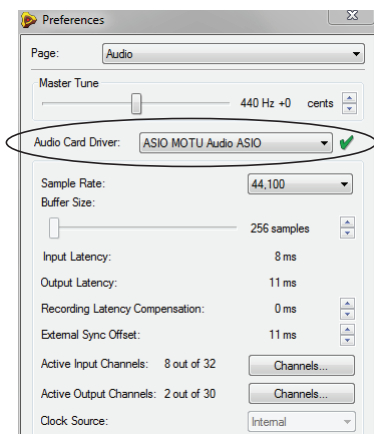


Figure 7-5: Enabling the MOTU Audio ASIO driver in Reason

Reaper

In Cockos Reaper, access the Preferences and click *Devices* under the *Audio* preferences. Choose *ASIO* for the audio system and choose *MOTU Audio ASIO* from the ASIO driver menu as shown below in Figure 7-6.

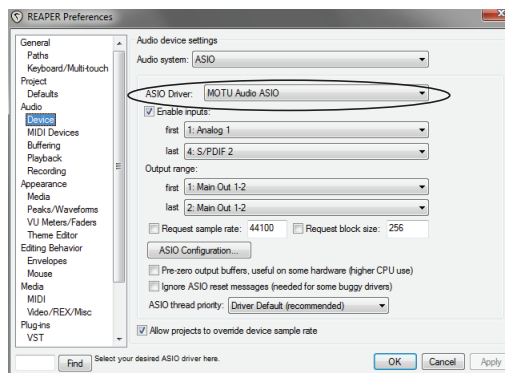


Figure 7-6: Enabling the MOTU Audio ASIO driver in Reaper

SONAR

In Cakewalk SONAR, you can use the MOTU ASIO driver or the MOTU WDM driver. Note that when using ASIO, only one audio interface may be selected for use in SONAR.

Using the MOTU ASIO driver

- 1 Go to SONAR's Preferences.
- 2 In the Audio preferences section, choose *Playback and Recording*.
- 3 Choose ASIO as the Driver Mode, as shown below.

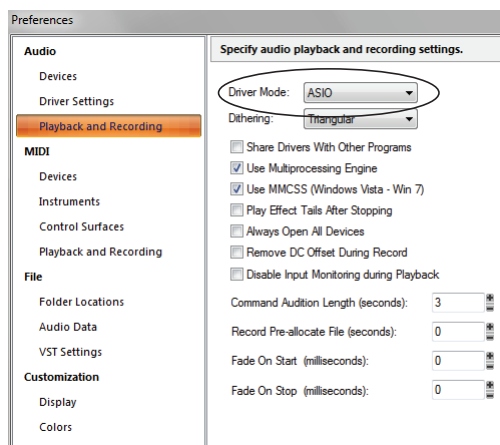


Figure 7-7: Enabling the MOTU Audio ASIO driver in SONAR.

- 4 Next, in the Audio preferences section, choose *Devices*.
- 5 Check the 896mk3 Hybrid inputs and outputs that you wish to use and uncheck the ones you don't, as shown in Figure 7-8 on page 61.

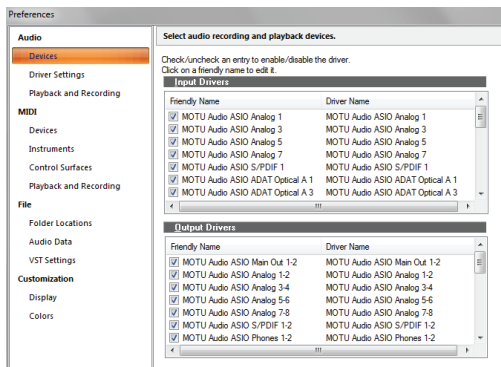


Figure 7-8: Enabling inputs and outputs in SONAR.

Using the MOTU WDM driver

1 Go to SONAR's Preferences.

2 In the Audio preferences section, choose *Playback and Recording*.

3 Choose *WDM/KS* as the Driver Mode, as shown below.

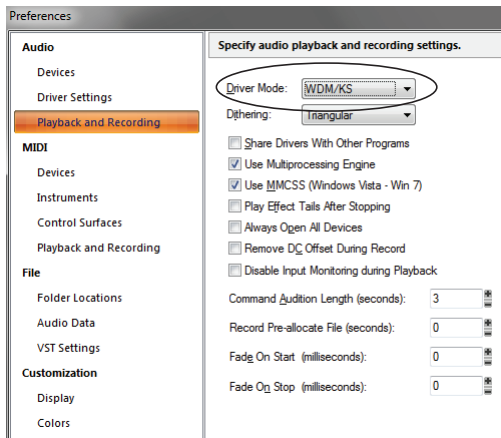


Figure 7-9: Enabling the MOTU WDM driver in SONAR.

4 Next, in the Audio preferences section, choose *Devices*.

5 Check the 896mk3 Hybrid inputs and outputs that you wish to use and uncheck the ones you don't, as shown in Figure 7-7 on page 60.

Other audio settings in SONAR

There are a few other settings in SONAR that impact 896mk3 operation:

1 In the Audio preferences section, choose *Driver Settings*.

2 Choose an 896mk3 input and output for recording and playback timing as shown below in Figure 7-10. It doesn't matter which input or output you choose.

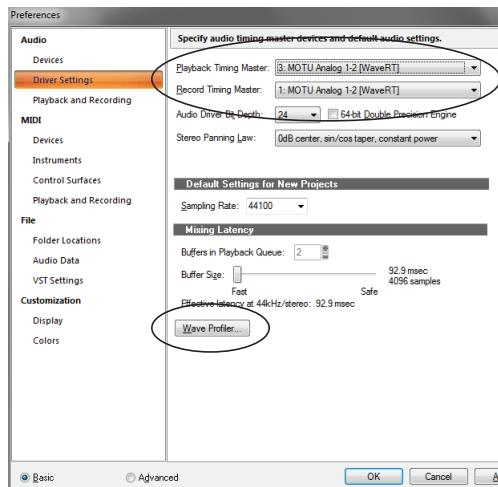


Figure 7-10: Make sure you have chosen a 896mk3 input and output for the playback and recording timing master settings.

3 Additionally, if you are using the MOTU WDM driver, click the *Wave Profiler* button and run the Wave Profiler. This process chooses the optimum settings for the 896mk3 hardware.

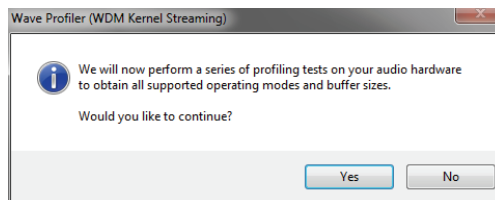


Figure 7-11: Running the Wave Profiler in Cakewalk's SONAR.

Sound Forge

In Sony Sound Forge, go to the Preferences window, click the *Audio* tab, and then choose *MOTU Audio* from the *Audio device type* menu.

Other audio software

For other audio applications, the procedure is similar to that shown above. Consult your owner's manual for further information.

REDUCING LATENCY

On Windows, audio I/O buffer size is handled by the audio driver rather than the host audio application.

In the MOTU Audio Console, *Samples Per Buffer* provides an adjustable audio buffer setting that lets you control the amount of delay you'll hear when monitoring live inputs through your host audio software or processing them with software plug-ins.

For information, see the "Adjusting the audio I/O buffer" section of chapter 8, "Reducing Monitoring Latency" (page 65).

WORKING WITH 896MK3 INPUTS AND OUTPUTS

Once you've enabled the 896mk3 Hybrid's driver, 896mk3 Hybrid audio inputs and outputs will appear in your host software wherever audio inputs and outputs are listed. If you don't see the optical inputs and/or outputs, check MOTU Audio Console to make sure they are turned on. If you don't plan to use either optical bank, turn it off to conserve computer bandwidth.

Remember that the main outs and headphone output can be configured in MOTU Audio Console to mirror other outputs. In addition, the 896mk3 Hybrid optical banks can be disabled or configured as either ADAT optical or TOSLink. This will affect how they appear (and if they appear at all) in your

host software's audio input lists. See "Optical input/output" on page 44, "Phones Assign" on page 43 and "Main Out Assign" on page 44.

AUDIO INPUT AND OUTPUT NAMES

The MOTU Audio driver supplies text string labels for the 896mk3 Hybrid's audio inputs and outputs to clearly identify each one, but some applications do not display these labels.

For information on how you can identify each input and output in a numbered list like this, see Appendix A, "Audio I/O reference" page (125).

NUMBER OF CHANNELS

If your host audio software requires that you specify the number of audio voices or channels you will be using, be sure to choose enough channels to cover the 28 inputs and 32 outputs provided by your 896mk3 Hybrid — although the number of channels may depend on how your 896mk3 Hybrid is configured.

PROCESSING LIVE INPUTS WITH HOST PLUG-INS

If you patch a live input (such as a MIDI synthesizer) through a plug-in effect in your host software, you might hear a slight delay. There are several ways to reduce this delay. For details, see chapter 8, "Reducing Monitoring Latency" (page 65).

WORKING WITH CUEMIX FX MIXING AND EFFECTS

The 896mk3 Hybrid provides powerful external mixing, EQ, compression and reverb, which you can operate hand-in-hand with your host's mixing environment. For example, the 896mk3 Hybrid can serve as a monitor mixer routing channels to musicians, or it can serve as an integrated extension of your host's mixing environment. If you program an 896mk3 Hybrid mixing and processing configuration that goes hand in hand with your host project, be sure to use the file save

features in CueMix FX to save the 896mk3 Hybrid settings as a file in your host project folder for instant recall of all settings. See chapter 9, “CueMix FX” (page 71) for complete details.

SYNCHRONIZATION

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 28 for the proper hardware connections. Use the synchronization diagrams to be clear about how you will be synchronizing your audio software and the 896mk3 Hybrid to the other components of your system.

Synchronizing digital audio connections

If you have devices connected to the 896mk3 Hybrid digital inputs (optical or RCA S/PDIF), you need to be concerned with the synchronization of the 896mk3 Hybrid’s digital audio clock with other devices connected to it digitally (if any). For example, if you have a digital mixer connected to the 896mk3 Hybrid via an ADAT optical light pipe cable, you need to make sure that their audio clocks are phase-locked. For details, see “Choosing a clock source for optical connections” on page 25 and “Making sync connections” on page 28. If you don’t have any digital audio devices connected to the 896mk3 Hybrid, digital audio phase-lock does not apply to you.

Resolving directly to time code (with no synchronizer)

If your host audio software supports ASIO2’s sample-accurate positioning protocol, then it can resolve to the 896mk3 Hybrid’s built-in time code synchronization feature. To resolve your 896mk3 Hybrid directly to SMPTE time code with no additional synchronization devices, use the setup shown in “Syncing to SMPTE time code” on page 34.

USING A FOOT SWITCH

Use a foot switch connected to the 896mk3 Hybrid to trigger recording punch-in and punch-out, or any other feature in your host audio software that is assigned to a keyboard shortcut.

To configure the keystrokes triggered by the foot switch, visit MOTU Audio Console. (See “Enable Pedal” on page 43.)

24-BIT OPERATION

Your 896mk3 hardware fully supports 24-bit recording capabilities in any audio software that supports it. Simply enable 24-bit operation as instructed by the software. The 896mk3 Hybrid always supplies the software a 24-bit data stream, and when you enable 24-bit operation, it simply uses all 24-bits supplied by the 896mk3 Hybrid 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 guitar input signal through an amp modeling plug-in running in your audio sequencer.

This delay is caused by the amount of time it takes for audio to make the entire round trip through your computer, from when it first enters an 896mk3 input, passes through the 896mk3 hardware into the computer, through your host audio software, and then back out to an 896mk3 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 896mk3's CueMix FX digital mixer to patch the input directly to your monitor outs via the 896mk3 audio hardware. The 896mk3 even provides effects processing (EQ, compression and reverb), which can be applied on input, output, or even at the bussing stage, just like a conventional mixer. For details, see "CueMix FX hardware monitoring" on page 68.

If you *do* need to process a live input with host software plug-ins, or if you are playing virtual instruments live through your 896mk3 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 896mk3: 1) through the computer or 2) via the 896mk3 CueMix FX hardware mixer.

Figure 8-1 shows method 1, which allows you to apply host-based effects processing 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 FX hardware-based monitoring, which lets you hear what you are recording with no monitoring delay and no computer-based effects processing. Instead, input

is routed directly to an output, either with or without 896mk3-based effects processing (EQ, compression or reverb). See “CueMix FX hardware monitoring” later in this chapter for details on how to use CueMix FX with your audio software, or with the included CueMix FX software.

If the material you are recording is suitable, there is a third way to monitor live input: use both methods (Figure 8-1 and Figure 8-2) at the same time. For example, you could route guitar to both the computer (for an amp model effect) and mix that processed signal on the main outs with dry guitar from CueMix FX — or perhaps with a touch of 896mk3 Classic Reverb.

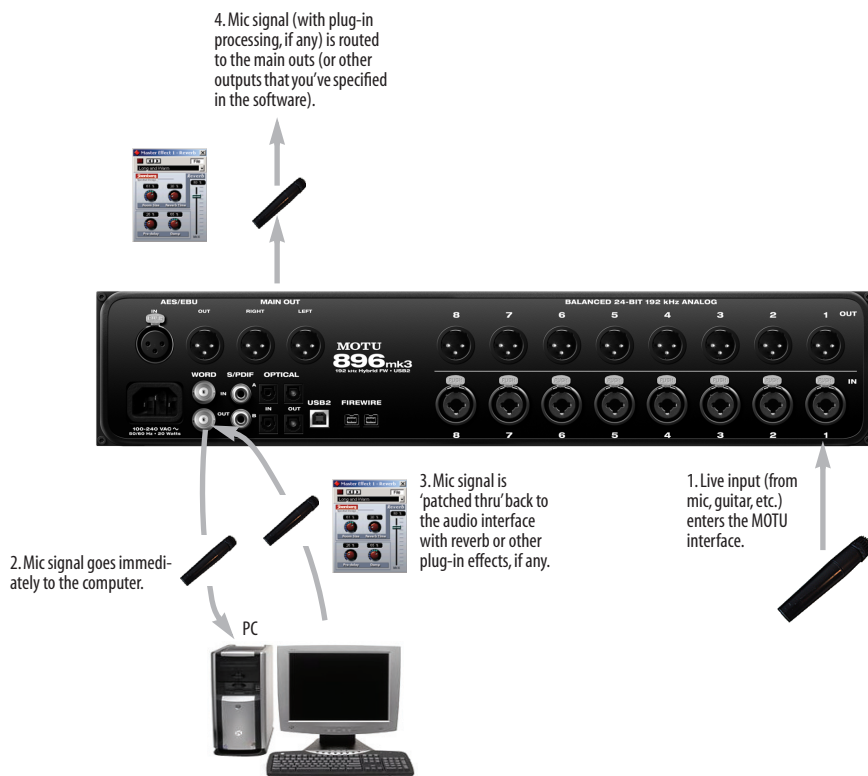


Figure 8-1: There are two ways to monitor live audio inputs with an 896mk3: 1) through the computer or 2) via CueMix FX hardware monitoring. This diagram shows method 1 (through the computer). When using this method, use your host software's buffer setting to reduce the slight delay you hear when monitoring the live input, but don't lower it too much, or your computer might get sluggish.

ADJUSTING THE AUDIO I/O BUFFER

A *buffer* is a small amount of computer memory used to hold data. For audio interfaces like the 896mk3, buffers are used for the process of transferring audio data in and out of the computer. The size of the buffers determines how much delay you hear when monitoring live inputs through your audio software: larger buffers produce more delay; smaller buffers produce less.

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

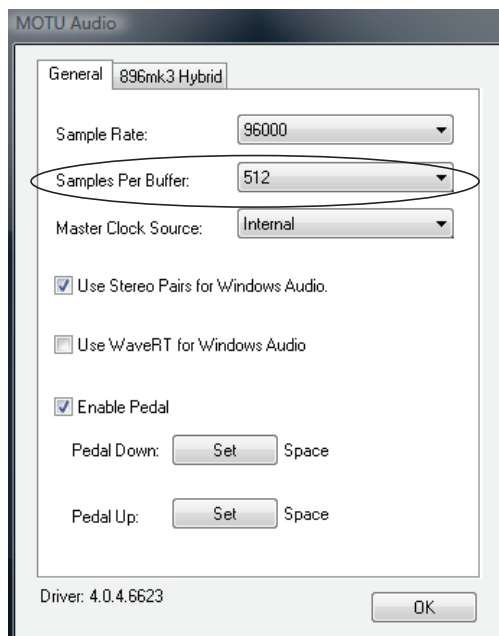


Figure 8-3: Lowering the 'Samples Per Buffer' setting in MOTU Audio Console reduces patch thru latency. But doing so increases the processing load on your computer, so keep an eye on the Performance Monitor in your host audio software.

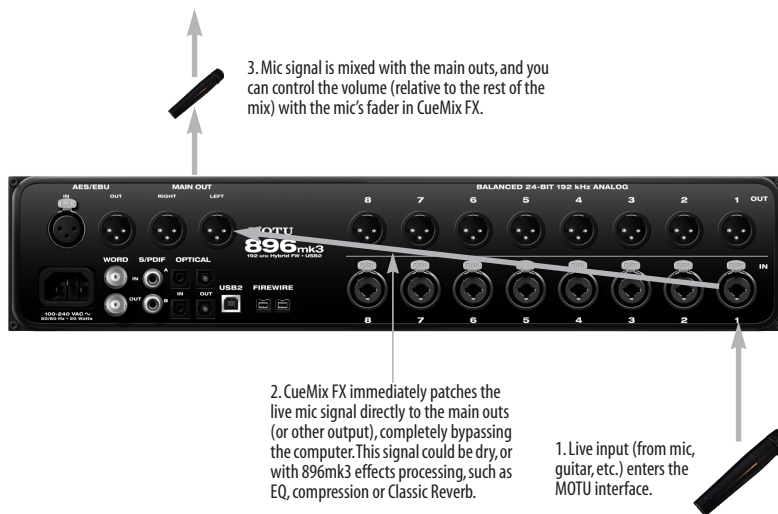


Figure 8-2: This diagram shows the signal flow when using CueMix FX no-latency monitoring. Notice that this method does not process the live input with plug-ins in your audio software. Instead, you can apply 896mk3 effects, such as the reverb, EQ and/or compression.

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 FX HARDWARE MONITORING

The 896mk3 has a more direct method of patching audio through the system. This method employs the 896mk3's *CueMix FX* digital mixer. When enabled, CueMix activates hardware patch-thru in the 896mk3 itself. CueMix FX 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 FX imposes no strain on the computer.

The trade-off, however, is that CueMix FX bypasses your host audio software. Instead, live audio inputs are patched directly through to outputs in the 896mk3 itself and are mixed with disk tracks playing back from your audio software. This means that you cannot apply host-based 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 FX 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 FX. Instead, reduce latency with the buffer setting (as explained earlier in this chapter).

TWO METHODS FOR CONTROLLING CUEMIX FX

There are two ways to control CueMix FX:

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

You can even use both methods simultaneously.

Using CueMix FX

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

CueMix FX allows you to create up to eight separate 896mk3 stereo mixes, or any other desired routing configurations. These routings are independent of your host audio software. For complete details, see chapter 9, “CueMix FX” (page 71).

Controlling CueMix FX from your audio software

Some ASIO-compatible audio applications, such as Cubase and Nuendo, allow you to control CueMix FX monitoring from within the application (without the need to use CueMix FX). In most cases, this support consists of patching an 896mk3 input directly to an output when you record-arm a track. Exactly how this is handled depends on the application.

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

Controlling CueMix FX from within Cubase or Nuendo

To turn on CueMix in Cubase or Nuendo, enable the Direct Monitoring check box in the Device Setup window.

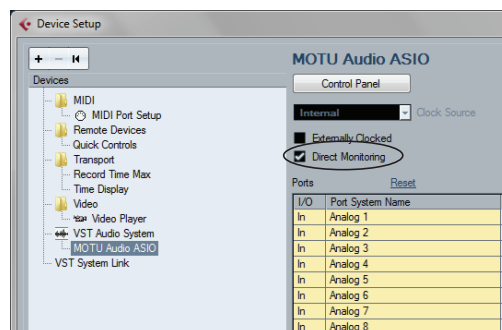


Figure 8-4: Activating the Direct Monitoring option in Cubase or Nuendo.

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 FX with WDM- or Wave-compatible software

Run CueMix Console (chapter 9, “CueMix FX” (page 71)) 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 FX

OVERVIEW

CueMix FX is a cross-platform software application that provides graphic, on-screen control for the 896mk3 Hybrid’s flexible CueMix FX on-board mixer and effects processing.

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

For information about programming the 896mk3’s on-board mixing and effects using the front panel LCD, see chapter 6, “Front Panel Operation” (page 47).

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A 16-BUS MIXER WITH EQ, COMPRESSION AND REVERB

All 896mk3 inputs and outputs can be routed to the on-board CueMix FX 16-bus (8 stereo bus) digital mixer driven by hardware-based DSP with 32-bit floating point precision.

The CueMix FX mixer allows you to apply no-latency effects processing to inputs, outputs or busses directly in the 896mk3 hardware, independent of the computer. Effects can even be applied when the 896mk3 is operating stand-alone (without a computer) as a complete rack-mounted mixer. Input signals to the computer can be recorded wet, dry, or dry with a wet monitor mix (for musicians during recording, for example).

Effects include:

- Classic Reverb with tail lengths up to 60 seconds
- 7-band parametric EQ modeled after British analog console EQs
- A standard compressor with conventional threshold/ratio/attack/release/gain controls
- The Leveler™, an accurate model of the legendary LA-2A optical compressor, which provides vintage, musical automatic gain control

The 896mk3's flexible effects architecture allows you to apply EQ and compression on every input and output (a total of 58 channels), with enough DSP resources for at least one band of parametric EQ and compression on every channel at 48 kHz. DSP resources are allocated dynamically and a DSP meter in the CueMix FX software allows you to keep tabs on the 896mk3's processing resources.

Each input, output and mix bus provides a send to the Classic Reverb processor, which then feeds reverb returns to mix busses and outputs, with a selectable split point between them to prevent send/return feedback loops.

ADVANTAGES OVER HOST-BASED MIXING AND PROCESSING

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

- CueMix has no buffer latency. Thanks to the 896mk3's DSP chip, CueMix provides the same throughput performance as a digital mixer.
- CueMix mixing and effects processing imposes 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 896mk3 to operate as a portable, stand-alone mixer with effects.

CUEMIX FX INSTALLATION

CueMix FX is installed with the rest of your 896mk3 software.

CUEMIX FX BASIC OPERATION

Here is a brief overview of the *CueMix FX* mixer.

Eight stereo mix busses

CueMix FX provides eight stereo mix busses: Bus 1, Bus 2, Bus 3, and so on. Each mix bus can take any number of inputs and mix them down to any 896mk3 output pair that you choose. For example, Bus 1 could go to the headphones, Bus 2 could go to the main outs, Bus 3 could go to a piece of outboard gear connected to analog outputs 7-8, etc.

Many inputs to one output pair

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

Viewing one mix bus at a time

CueMix FX displays one mix bus at a time in the *Mixes* tab (Figure 9-2 on page 75). To select which mix you are viewing, choose it from the mix bus menu (Figure 9-2). The mix name appears above the mix bus master fader (Figure 9-2), where you can click the name to change it.

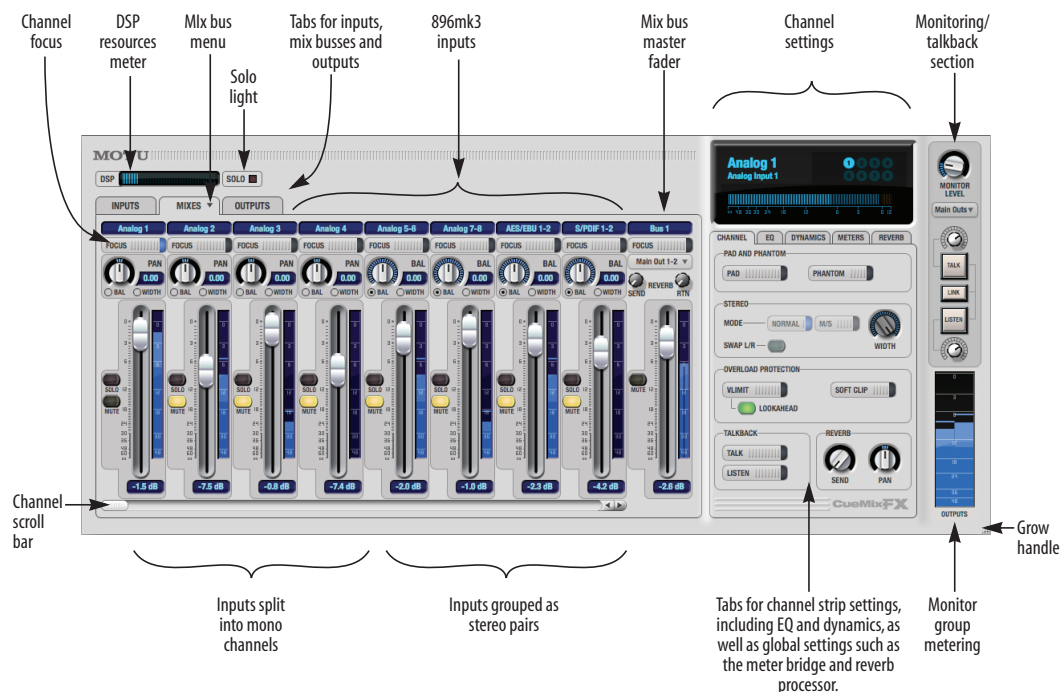


Figure 9-1: *CueMix FX* is a virtual mixer that gives you control over the 896mk3's on-board mixing features.

Each mix bus is independent

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

Input channels

The *Inputs* tab (Figure 9-3 on page 77) gives you access to settings for individual 896mk3 inputs (or input pairs), such as phase, trim, EQ and dynamics processing. Each input also includes a send to the 896mk3's global reverb processor. These settings are applied to the signal before it goes anywhere else (to a mix bus or the computer).

Output channels

The *Outputs* tab (Figure 9-6 on page 80) gives you access to settings for each 896mk3 output pair, including EQ, dynamics processing and send/return controls for feeding and returning the output signal to/from the 896mk3's global reverb processor. These settings are applied to the signal just before it is sent to the output.

Channel focus and settings

Click the focus button for a channel (Figure 9-1) to view channel-specific parameters in the Channel Settings section of the CueMix FX window (Figure 9-1). Separate tabs are provided for channel-specific settings (channel strip, EQ and dynamics), plus the global meter bridge and reverb processor.

Global reverb processor

The 896mk3 has a global reverb module (Figure 9-23 on page 94). Once it has been activated, you can feed signals to the reverb processor from various points in the 896mk3 mix matrix via input sends, bus sends and output sends. Stereo output from the reverb processor can then be fed back to mix busses or output pairs using reverb returns.

Other features

CueMix offers many additional features, discussed in this chapter, such as talkback/listenback, extensive metering, graphic editing of certain effects parameters, monitor grouping and more.

Widening the CueMix FX window

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

THE MIXES TAB

Click the *Mixes* tab (Figure 9-2) to gain access to the 896mk3's eight stereo mix busses. The Mixes tab displays one mix bus at a time.

Viewing a mix

Choose the mix you wish to view from the mix bus menu (in the Mixes tab itself, as shown in Figure 9-2). The menu shows all mixes by name, followed by the 896mk3 output pair to which each bus master fader is currently assigned, if any.

Naming a mix

Click the mix name at the top of the mix bus master fader (Figure 9-2) to edit the name.

Assigning a mix bus output

Choose the desired output pair for the mix bus from the bus output menu (Figure 9-2). The bus output menu displays all current available (enabled) 896mk3 output pairs. If a bus is already assigned to an output pair, the bus name appears next to the output pair name to indicate that the output pair is already taken by a bus. Only one bus can be assigned to any given output pair. If you choose an output already assigned to another bus, that bus output will become disabled.

Bus fader

The bus fader (Figure 9-2) 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.

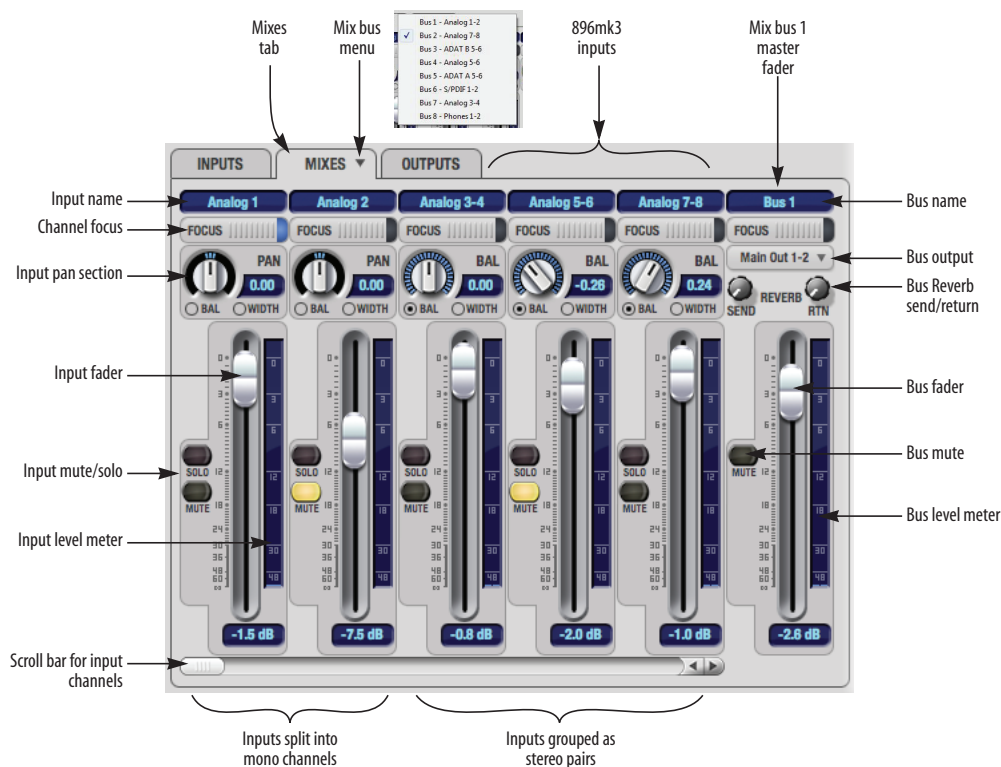


Figure 9-2: The Mixes tab.

Bus mute

The bus mute button (Figure 9-2) disables (silences) the mix.

Bus level meter

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

Bus reverb send/return

The bus reverb send (Figure 9-2) feeds the output of the mix bus, pre-fader, to the 896mk3's global reverb processor, where it is merged with any other signals being fed to the reverb. The reverb's output can then be fed back into the mixer at various return points, including the bus return (discussed below).

The bus reverb *return* (Figure 9-2) feeds the output of the 896mk3's global reverb processor into the mix bus, pre-fader. This includes any other signals currently being fed to the reverb. The bus reverb return is disabled (grayed out) when the reverb *Split Point* is set to *Output* to eliminate the possibility for feedback loops created by reverb send/return loops. See "Split point" on page 94.

Input section

The horizontally scrolling area in the Mix tab to the left of the master fader (Figure 9-2) displays channel strips for all currently enabled 896mk3 inputs.

Input channel focus

Click the channel focus button (Figure 9-3) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 9-7 on page 81). Clicking the mix bus master fader focus button brings the assigned output into focus, if there is one. Channel focus also determines which channels are being scoped by CueMix's audio analysis tools, as explained in "Choosing channels for audio analysis" on page 99.

Input pan section

The input pan knob (Figure 9-2) pans the input across the bus stereo outputs. If the input itself is grouped as a stereo pair (in the Inputs tab), two forms of panning control are provided:

Balance

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

Width

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

Input fader and mute/solo

To add an input to a mix, or remove it, click its Mute button. To solo it, use its Solo button. Use the input fader (Figure 9-2) to adjust the level for the input in the mix. Note that an input can have different level, pan, mute and solo settings for different mixes. Input channel level meters are post-fader.

If any solo button on the current (active) bus is enabled, the Solo Light (Figure 9-1) will illuminate.

THE INPUTS TAB

The 896mk3 provides many features for managing analog and digital input signals. Some of these features, such as the 896mk3's digitally controlled analog trims, are implemented in the analog domain; others are implemented in the digital domain as DSP applied to the digital signal (after the A/D converter on analog inputs). Click the *Inputs* tab (Figure 9-3) to access and control all of these input channel settings for each 896mk3 input or input pair.

Input tab settings are global

Except for the reverb send, all settings you make in the Input tab are applied to the input signal before it goes anywhere else (to a mix bus or the computer). For example, if you apply EQ and compression to the input signal, you will record the processed version of the signal in your host audio software running on the computer. If you need to

record a completely unprocessed input signal, do not apply any changes to it in the Input tab. The only exception to this is the reverb send, which simply splits the input signal and feeds a copy of it to the 896mk3's reverb processor.

Signal flows from top to bottom

Settings in each Input tab channel strip are generally applied to the signal in order from top to bottom. Input channel signal flow is as follows: front-panel trim, overload protection, phase, stereo versus M/S decoding, width, L/R swap, EQ, dynamics and reverb send.

Input name

The input name appears at the top of the input channel. This name also appears in host audio software on the computer (if the software supports channel names).

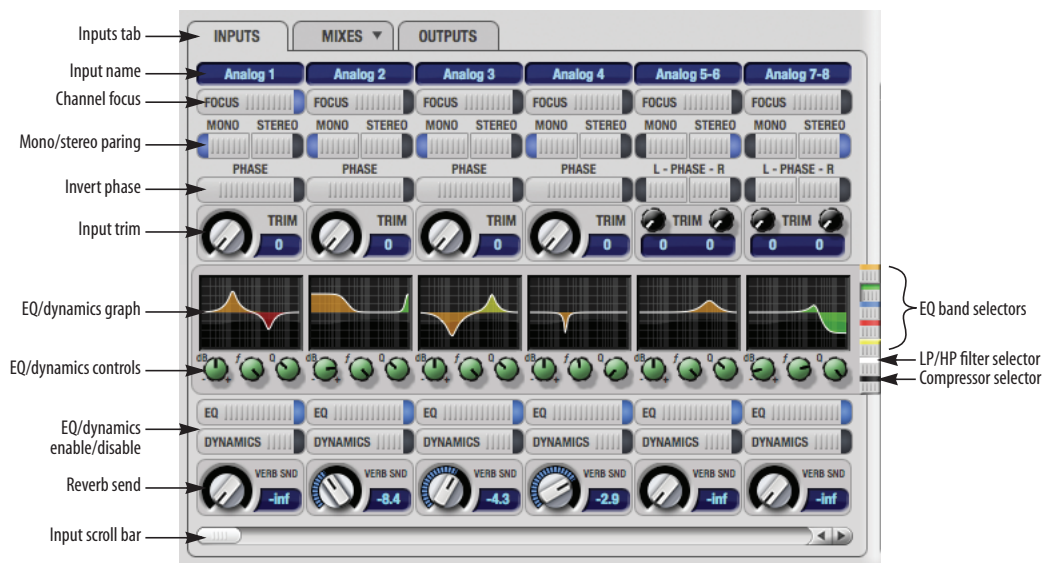


Figure 9-3: The Inputs tab.

Input channel focus

Click the channel focus button (Figure 9-3) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 9-7 on page 81).

Mono/stereo pairing

Click the *Mono* button (Figure 9-3) if you would like an input to be treated as a mono channel. If you would like to work with it as one channel of a linked stereo pair, click the *Stereo* button. Inputs are grouped in odd/even pairs (mic 1-2, Analog 1-2, 3-4, etc.) Stereo pairs appear as a single channel strip in the CueMix FX mixer (in all tabs).

Invert phase

Click the *Phase* button (Figure 9-3) to invert the phase of the input signal. For stereo pairs, you can invert the phase for the left and right channels independently.

Input trim

All 896mk3 inputs, both analog and digital, offer continuously variable input trim. In all cases, trim level can be controlled digitally in 1 dB increments. This includes the digitally controlled analog trims on the eight analog inputs. Here is a summary of input trim ranges for each type of 896mk3 input:

Input	Trim cut	Trim boost	Trim Range
Analog inputs	0 dB	+53 dB	53 dB
AES/EBU	0 dB	+12 dB	12 dB
S/PDIF (RCA)	0 dB	+12 dB	12 dB
ADAT optical	0 dB	+12 dB	12 dB
S/PDIF (TOSLINK)	0 dB	+12 dB	12 dB

Once you adjust the trim levels, you can save them as a file on disk for future instant recall. See “Saving and loading hardware presets” on page 98 and “Configurations menu” on page 116.

Input EQ and dynamics

The 896mk3 lets you apply 7-band parametric EQ and dynamics processing (DSP) to any input, analog or digital.

The controls in the EQ/Compression section of the Inputs tab (Figure 9-3) let you edit EQ and compression settings within the context of the channel strip. This is ideal when you are comparing settings among neighboring channels, or perhaps even applying the same setting across all inputs. However, for more detailed editing of EQ and compression settings for an input channel, you can click its Focus button and view the settings in the Channel Section of the CueMix FX window (Figure 9-1). This section even provides graphical editing of EQ curves and the compressor graph, allowing you to click and drag directly on the graphic. For details see “The channel settings section” on page 81.

The EQ/Dynamics graph

The EQ/Dynamics graph for each input channel strip (Figure 9-3) provides a thumbnail view of the EQ curves or Compressor graph for the channel. This graphic is for display purposes only; it cannot be edited directly. To change the EQ settings in this graph, use the two or three knobs below, as explained in the following sections. If, however, you would like to edit the EQ curves graphically, you can do so in the EQ tab (Figure 9-10 on page 84).

EQ/Dynamics selectors

The EQ/Dynamics selector buttons along the right-hand edge of the EQ/Dynamics section (Figure 9-3) allow you to choose what you are viewing and editing in the EQ/Dynamics section.

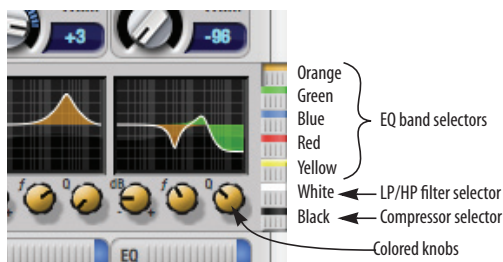


Figure 9-4: The EQ/Dynamics selectors.

Click the selector (Figure 9-4) for the desired EQ band, low-pass (LP) filter, high pass (HP) filter or compressor to view it across all channels.

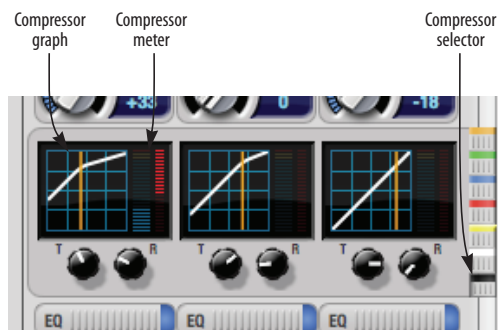


Figure 9-5: The Compressor controls.

🔑 **Shortcut:** hold down the Option/Alt key while clicking an EQ selector button to show just that band of EQ in the graphs. Click any selector again to return to viewing all bands.

Using the EQ/Dynamics knobs

Once you have chosen the desired EQ band, or the compressor, you can modify its settings using the two or three knobs below the graph. The knobs match the color of the currently selected effect, to help remind you of which effect you are currently editing.

🔑 **Important:** before you can modify the settings of an EQ band using the three knobs below the graph, the EQ band must be enabled. This is done in the EQ tab (Figure 9-10), as explained in “Enabling EQ” on page 84.

EQ/Dynamics enable/disable buttons

Click the EQ or Dynamics button at the bottom of the input channel (Figure 9-3) to toggle the effect on or off. Note that you can program EQ and compressor settings, even when the effect is currently disabled. (You just won’t hear the result until you enable it.)

Reverb send

The input reverb *send* (Figure 9-3) feeds the input signal to the 896mk3’s global reverb processor, where it is merged with any other signals being fed to the reverb. The reverb’s output can then be fed back into a mix or output pair. The send occurs after all other settings in the input channel strip (phase invert, EQ, compression, etc.)

THE OUTPUTS TAB

The Outputs tab (Figure 9-6) lets you apply EQ, dynamics and reverb to any output pair, just before the signal leaves the 896mk3. This is processing that occurs at the very end of the signal flow, after everything else (host based effects, 896mk3 input or bus processing, mixing, and so on). Processing is done in the digital domain, just before the signal goes analog through the D/A converter. Output tab processing is applied to the entire output mix (all signals being mixed to the output from various sources).

Signal flows from top to bottom

Settings in each Output tab channel strip are applied to the signal in order from top to bottom. For example, EQ occurs before Dynamics, which is applied before the reverb send and return.

Output name

The output name appears at the top of the input channel. Output names are global and will also appear in host audio software on the computer (if the software supports channel names).

Output channel focus

Click the channel focus button (Figure 9-6) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 9-7 on page 81). Channel focus also determines which channels are being scoped by CueMix’s audio analysis tools, as explained in “Choosing channels for audio analysis” on page 99.

Output EQ and Dynamics

The EQ/Dynamics section in the Outputs tab (Figure 9-6) works identically to the EQ/Dynamics section for the Inputs tab (Figure 9-3). See “Input EQ and dynamics” on page 78.

Output reverb send/return

The output reverb *send* (Figure 9-6) feeds the signal for the output to the 896mk3’s global reverb processor, where it is merged with any other signals being fed to the reverb. The reverb’s output can then be fed back into the mixer at various return points, including the same output from which it was sent (discussed below). The output reverb

send is disabled (grayed out) when the reverb *Split Point* is set to *Mix* to eliminate the possibility for feedback loops created by reverb send/return loops. See “Split point” on page 94.

The output reverb *return* (Figure 9-6) feeds the output of the 896mk3’s global reverb processor directly to the output. This includes any other signals currently being fed to the reverb.

Both the send and return occur after EQ and dynamics, but before listenback and talkback.

Talkback/Listenback

Click the *Talkback* or *Listenback* buttons (Figure 9-6) to toggle whether the output pair is included in the Talkback or Listenback group. See “Talkback and listenback” on page 96.

Monitor group assign

Click the Monitor buttons (Figure 9-6) to toggle whether the output pair is included in the Monitor group. See “The Monitor Group” on page 95.

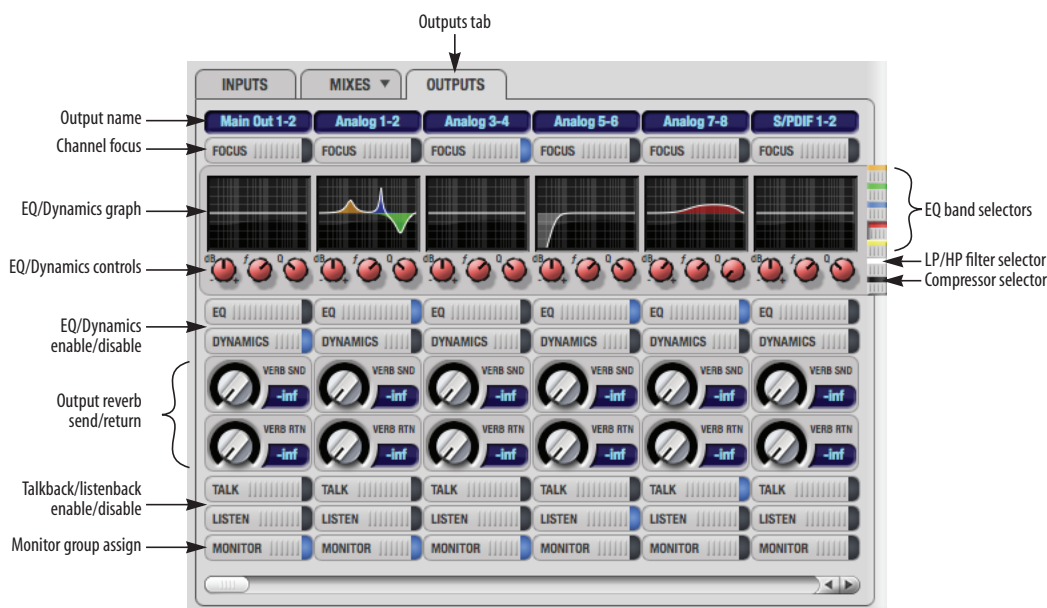


Figure 9-6: The Outputs tab.

THE CHANNEL SETTINGS SECTION

The channel settings section in the CueMix FX window (Figure 9-1) displays three tabs for Channel, EQ and Dynamics settings for the channel with the current focus. There are also two global tabs: the Meter Bridge and the Reverb Processor, as shown below.

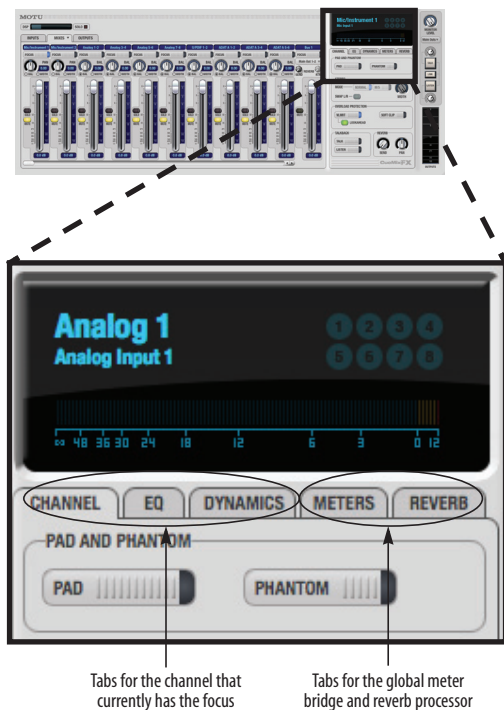


Figure 9-7: The Channel Settings section.

The Channel tab

The Channel tab (Figure 9-8) displays settings for input channels. Click any focus button in the Inputs tab to view the Channel tab settings for the channel.

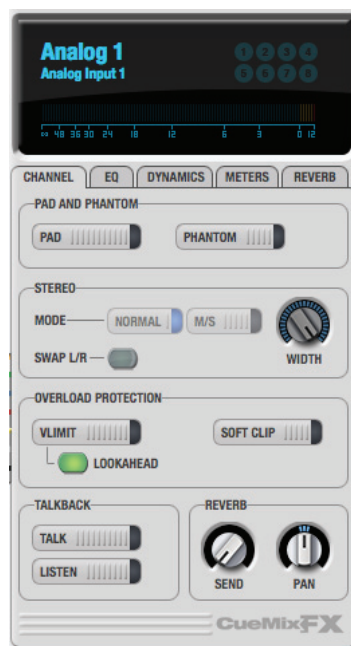


Figure 9-8: The Channel tab.

Signal flow

Settings in the Channel tab occur just before the EQ, dynamics and reverb sends in the Input tab channel strip (Figure 9-3 on page 77). Input channel signal flow is as follows: trim, overload protection, phase, stereo versus M/S decoding, width, L/R swap, EQ, dynamics and reverb send.

Pad and phantom

The Pad and phantom settings become active (ungrayed) when the focus is on a mic input (or mic input pair). These are the same -20 dB pad and 48V phantom power settings that you can also control from the 896mk3 front panel TRIM encoders. There are separate settings for each mic input.

Stereo settings

Inputs that have been grouped as stereo pairs in the Inputs tab (Figure 9-3) provide two stereo modes (Figure 9-8): *Normal* and *M/S*. *M/S* mode provides decoding for a mid-side microphone configuration.

The *Width* knob (Figure 9-8) provides control over the stereo imaging, going from a full stereo image to mono (both channels panned equally). See “Width” on page 76.

The *Swap L/R* button (Figure 9-8) lets you switch the left and right channels.

Overload protection

The Overload Protection section (Figure 9-8) provides two features that help prevent digital clipping on the eight analog inputs.

V-Limit™ (Figure 9-8) is a hardware limiter that helps prevent digital clipping from overloaded input signals. With *V-Limit* engaged, signals can go above zero dB (with limiting applied) to as high as +12 dB above zero with no distortion due to digital clipping. Click the *Lookahead* option for even better protection against sharp transients.

👁 *Lookahead* adds a 1.5 millisecond delay to the input signal. Therefore, when you enable lookahead for one or more inputs, all analog inputs (even ones without the limiter and lookahead enabled) are delayed together by 1.5 ms so that all eight inputs remain perfectly in phase with each other.

Additional or alternative protection can be applied by enabling *Soft Clip* (Figure 9-8). When enabled, *Soft Clip* engages just before clipping occurs and helps further reduce perceptible distortion. When *V-Limit* is turned off, *Soft Clip* begins to engage around -2 dB (just below zero); when *V-Limit* is turned on, *Soft Clip* begins to engage around +10 dB (just below +12 dB).

Talkback section

Click the *Talkback* or *Listenback* button (Figure 9-8) to toggle whether the input is the Talkback or Listenback input. Only one input can be the talkback input, and only one input can be the listenback input. See “Talkback and listenback” on page 96.

Reverb section

The *Send* in the reverb section (Figure 9-8) is the same control as the reverb send in the Input tab channel strip (Figure 9-3). See “Reverb send” on page 79. If the input is currently not grouped as a stereo pair in the Input tab (it is operating as a mono input), use the reverb *Pan* knob (Figure 9-8) to pan the mono signal for the stereo reverb processor.

Input meter and bus activity LEDs

When the Channel tab is active (Figure 9-8), the display above the tab provides a horizontal level meter and eight *bus activity LEDs* (Figure 9-9).

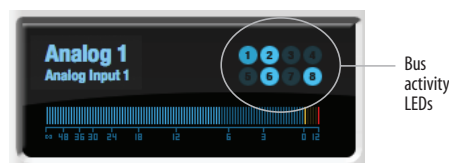


Figure 9-9: Input meter and bus activity LEDs.

The *input level meter* (Figure 9-9) is the same as the input meters in the Meters tab (Figure 9-22 on page 93) with the *Pre FX* button engaged, which shows the input level on the physical input itself, before any processing of any kind occurs within the 896mk3. This meter gives you the most accurate reading of the actual signal level hitting the input, regardless of any other settings (such as *V-Limit*, *Soft Clip* and so on). The clip indicator, however, happens after *V-Limit* and/or *Soft Clip*. This allows you to see when clipping occurs, even with these overload protection features engaged.

The *Bus Activity LEDs* (Figure 9-9) show you which mix busses the input signal is being fed to. For example, LED #6 will glow under the following conditions: the input is unmuted in mix bus 6, its fader is up, and there is signal activity from the input going into the mix bus.

The EQ tab

The EQ tab (Figure 9-10) displays the EQ settings for the input or output channel that currently has the focus. Click any focus button in the Inputs or Outputs tab to view the EQ tab settings for the channel.

Vintage EQ

Inspired by legendary British large console EQs, the 896mk3 *Vintage EQ section* (Figure 9-10) gives you the look, feel and sound of the most sought-

after classic equalizers. Five bands of center frequency parametric EQ filtering are provided, each with four EQ types that provide current popular EQ styles and vintage analog EQ styles alike. Two bands include shelf filtering. Two additional bands of variable slope low pass and high pass filtering are provided. The filter response display provides comprehensive control and visual feedback of the EQ curve being applied. With 64-bit floating point processing, the 896mk3 Vintage EQ has been carefully crafted and meticulously engineered to produce musical results in a wide variety of applications.

Enabling EQ

Each input and output channel has a global *EQ enable/disable* button (Figure 9-3 and Figure 9-6). This button enables or disables all bands of EQ for

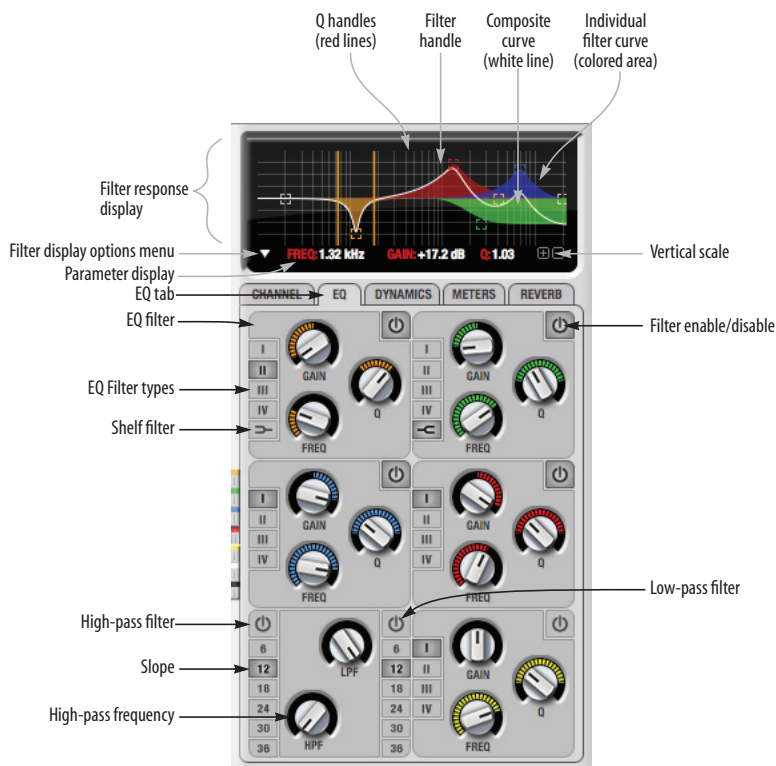


Figure 9-10: The EQ tab.

the channel. In addition, each individual band of EQ has a *Filter enable/disable* switch (Figure 9-10), allowing you to enable as few or as many bands as needed for each individual channel.

Vintage EQ Quick reference

Filter response display: Shows the response curve for the current settings.

Vertical scale: Lets you zoom the vertical scale of the filter response display.

Parameter display: Shows the precise numbers of the parameter you are adjusting (or hovering over with the arrow cursor). The labels (*frequency*, *gain*, etc.) match the color of the filter being displayed. When a filter handle is not selected and when the cursor is not hovering over the display, the parameter display shows the name of the current channel being edited (the channel that currently has the focus), as shown below:

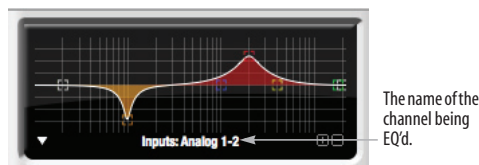


Figure 9-11: When a filter handle is not selected and when the cursor is not hovering over the display, the parameter display shows the name of the current channel being edited (the channel that currently has the focus).

EQ filter: one of five center bands of EQ that can be independently enabled and programmed.

Filter type: Lets you choose from one of four or five EQ styles for each independent band of EQ.

Low/High Pass filter: Both a low pass and high pass filter are supplied with six different slope settings.

Slope: Lets you choose the slope (fall off) characteristics of the low pass and high pass filter.

Q handle: Drag the Q handle lines to graphically adjust the Q setting for the currently selected filter. To select the filter, click its filter handle.

Filter handle: Drag this handle to graphically adjust the filter's boost/cut and/or frequency.

Composite curve (white line): shows the overall response curve of the current settings in the window.

Individual filter curve: Each filter has a color (indicated by its knobs). When filter curves are being displayed (the filter curve option is turned on), each individual filter's response curve is displayed in the filter's color.

Filter display options menu: Provides several options for controlling the filter display.

Filter enable/disable: Turns the filter on or off.

How the vintage EQ works

The Vintage EQ operates like a standard EQ filter, but with much more sophisticated processing algorithms “under the hood”. There are five bands of EQ, each with their own unique knob color, plus additional low pass and high pass filters. Each filter can be set to any center frequency you wish.

Each filter can be independently turned on or off with the enable/disable button (Figure 9-10). Each filter can be set to one of four different filter types (I, II, III or IV). The two top-most filters (orange and green) provide an extra low and high shelf setting, in addition to the four standard band settings. The additional low pass and high pass filters (lower left) have gray cutoff frequency knobs and six settings for slope (in octaves/dB).

Frequency response display

The frequency response display at the top of the window displays the response curve of the current settings in the window. The (horizontal) frequency range is from 10 hertz to 20 KHz. The (vertical)

amplitude scale is in dB and is adjustable between 3 and 24 dB using the *vertical scale* buttons (Figure 9-10).

Showing and hiding filter curves

To view a filter in the display, turn on the filter. The shape of the filter, according to its current settings, is shaded in the same color as the filter's knob(s). Use the *filter display options menu* (Figure 9-10) to show or hide them in the display.

Adjusting filters in the display

Each filter has a handle, displayed as shown below in Figure 9-12 (in the filter's color), for adjusting its boost/cut and/or frequency:

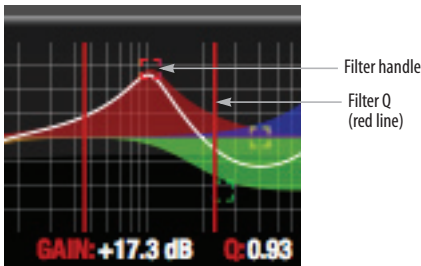


Figure 9-12: Drag the filter handle to adjust its frequency and/or boost/cut. Drag the Filter Q handles to adjust the Q.

For the EQ filters, when you click the handle, you'll also see lines on either side for adjusting the Q parameter, as shown above.

☞ **Shortcut:** hold down the Option/Alt key while clicking an EQ filter handle to enable or bypass that EQ filter band.

EQ filters

The EQ filters have three parameters:

Control	unit	range
Gain	dB	-20.00 to +20.00
Frequency	Hertz	20 Hz to 20 kHz
Q	n/a - see note below	0.01 to 3.00

Q

The Q setting does not have a unit of measurement. Rather, it is the ratio of the filter's center frequency to the bandwidth of the filter. In addition, the actual Q value for the EQ curve being applied is dependent on three factors: the gain setting, the filter style, and the Q setting.

Filter types

Each filter can be independently set to one of four different filter types: I, II, III and IV. These, and the additional shelf filters for the LMF and HMF band, are discussed in the section "EQ filter styles".

Returning to zero (or nominal frequency)

To return a knob to zero, or it's nominal frequency, double-click it.

EQ filter styles

EQ is one of the most widely used processing tools and can be applied to many different situations, from minor corrective tasks to highly creative applications. Over the years, many EQs have been engineered for specific applications or to achieve a certain sound. The Vintage EQ has been designed to be flexible enough to cover a broad range of applications. To that end, several different filter types are supplied, varying mostly in the way they handle the dynamic interaction between Gain and Q. This crucial relationship has been modeled to emulate the smooth and musical character of classic analog EQ circuits, in which the Gain/Q dependency was dictated by the actual circuit design and electrical components used. The following sections describe the character of each type of EQ filter and their suggested applications. In the three example illustrations (Figure 9-13 through Figure 9-16), the settings for the curves are the same for comparison purposes:

- Frequency = 1.00 kHz
- Q = 1
- Gain = +3, +10 and +20 dB

Type I

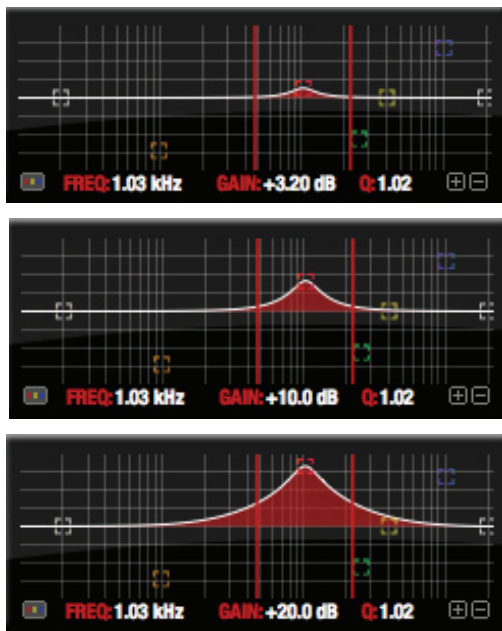


Figure 9-13: Type I EQ filter style.

The *Type I* EQ filter has the least amount of Gain/Q interaction, providing the most precision and control of all the EQ filter types. Even small adjustments in gain or reduction produce relatively high Q. This EQ style is best for situations that call for precise EQ adjustments requiring the maximum amount of individual parameter control. For more general shaping (e.g. full mixes) or subtle control (e.g. vocals), the other styles discussed in the following sections might be more appropriate. This filter type is the most similar to a standard parametric EQ.

Type II

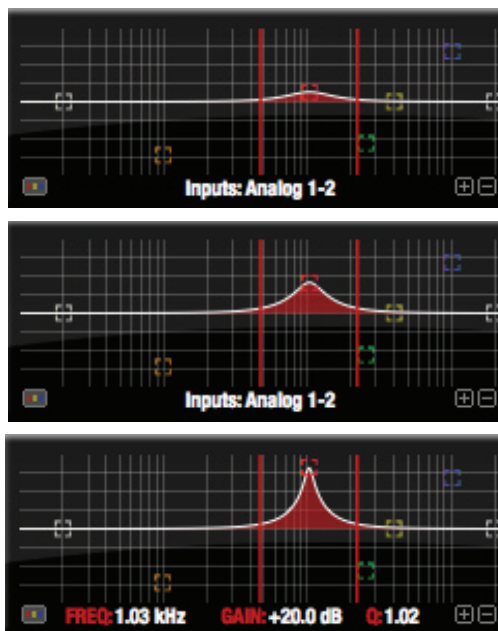


Figure 9-14: Type II EQ filter style.

The *Type II* EQ filter produces constant Q response during boost or cut. The Type II style emulates several classic legacy EQs and produces good results for resonance control on drums and percussion because it provides relatively high Q values with more extreme gain or cut settings.

Type III

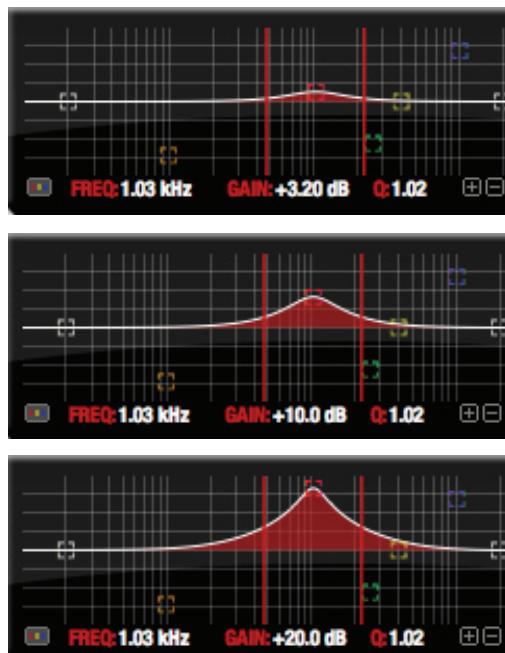


Figure 9-15: Type III EQ filter style.

The *Type III* EQ filter increases Q as boost is applied. Therefore, lower amounts of boost provide a softer, “wider” EQ effect (since the affected frequency range widens), while higher boost tends to sound louder and more “up front”, due to the increase in Q as the gain is increased. The more gentle Q curve at lower settings is well suited for overall EQ fills and more subtle corrections in instrument and vocal sources. Boosting or cutting by small amounts will seem to produce the effect that your ear expects, without the need to adjust Q. As a result, this filter style, and similar EQs with this characteristic behavior, are often referred to as being more “musical”. More specifically, this style emulates the classic Neve EQs, their modern derivatives and later SSL G series EQs. Many current popular outboard “boutique” EQs exhibit this same gain/Q relationship.

Type IV

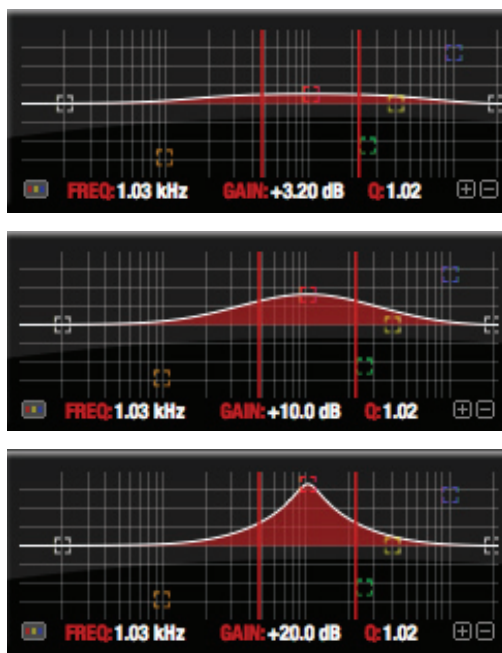


Figure 9-16: Type IV EQ filter style.

The *Type IV* EQ filter is a more extreme form of the *Type III* filter. It exhibits a high degree of interaction between Q and gain in order to maintain as closely as possible an equal amount of area under the response curve as gain is adjusted. Type IV is the most gentle of the four EQ styles and is ideal for large scale EQ adjustments, especially on sub-mixes and complete mixes. This EQ style is also ideal for any applications where subtle changes in the overall character of the sound are desired. For example, it can be used for mastering applications, such as the overall adjustments that must often be applied to entire tracks to match other tracks on the album.

Shelf filters

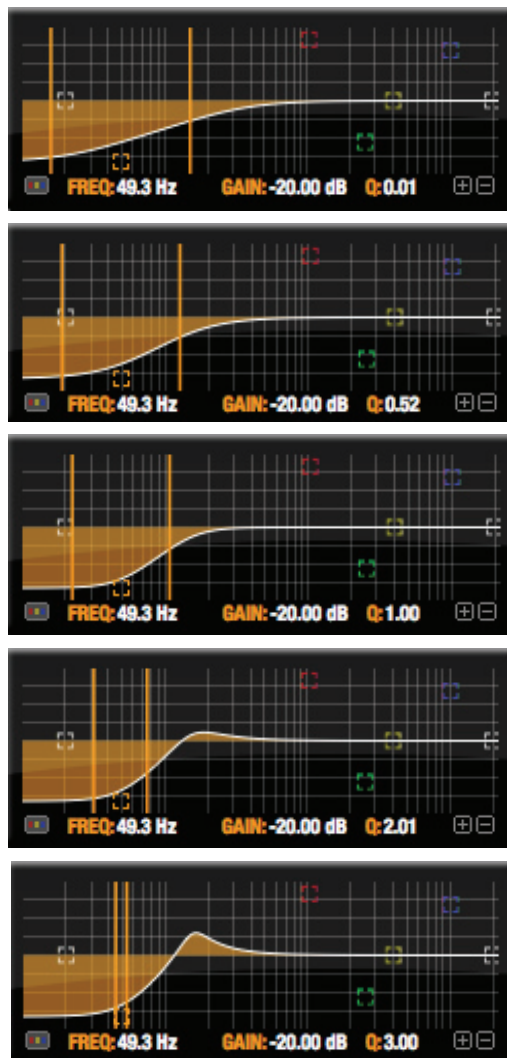


Figure 9-17: Shelf filter Q parameter overshoot.

When two top-most bands in the EQ tab are set to their *shelf filter* setting (Figure 9-10), the Q parameter controls the amount of overshoot applied to the response curve, as illustrated in Figure 9-17. When $Q = 0.01$ (the lowest setting), normal shelving is applied with no overshoot. This produces the response provided by a first order shelf. When $Q = 1.0$ (the default setting), the

response corresponds to a second order shelf, still with no overshoot. This is the same response as conventional parametric EQs. In some situations, this form of accurate, clean shelving can sound harsh, especially when compared to legacy analog EQs. To soften the results, the overshoot is increased as Q is increased, as shown Figure 9-17 for Q values of 1.00, 2.00 and 3.00. This overshoot region produces a boost in frequencies just above the cutoff, which compensates in a smooth, more pleasing fashion for the perceived drop in low frequencies being cut.

Conversely, when shelving boost is being applied, overshoot cuts frequencies just above the cutoff to again compensate in a smooth and pleasing fashion for the perceived boost in low frequencies:

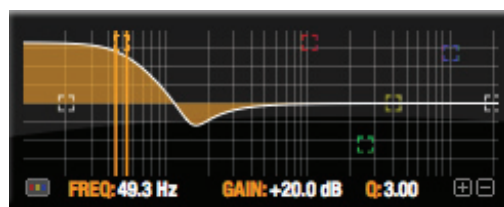


Figure 9-18: Overshoot when low shelf boost is applied.

Overshoot is also applied to high shelf boost and cut:

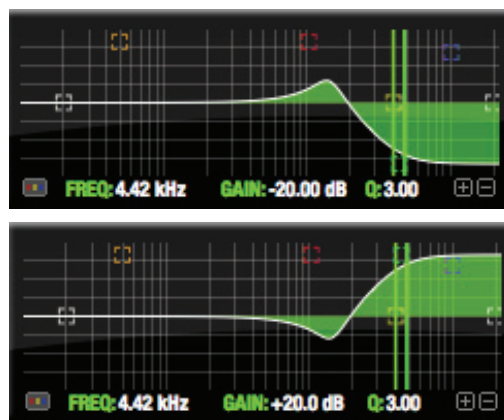


Figure 9-19: Overshoot when high shelf cut and boost is applied.

Overshoot tends to produce more of what one would expect to hear when applying shelving and is therefore considered to be more musical than shelving without overshoot. This effect, which has gained tremendous popularity among audio engineers, was first made popular in original Neve series EQs and later in the SSL G series.

At maximum the maximum Q setting of 3.00, the overshoot peaks at half the total boosted (or cut) gain. For example, with a maximum gain setting of +20dB, the loss in the overshoot region is -10 dB.

Overshoot curves are symmetrical for both cut and boost.

Low pass and high pass filters

The Vintage EQ low and high pass filters are similar to those found in most conventional parametric EQs (which usually have a fixed slope of 12 dB per octave), except that Vintage EQ provides six different slope (roll off) settings: 6, 12, 18, 24, 30 and 36 dB per octave. This control over the shape of the “knee” gives you a great deal flexibility and control for a wide variety of applications.

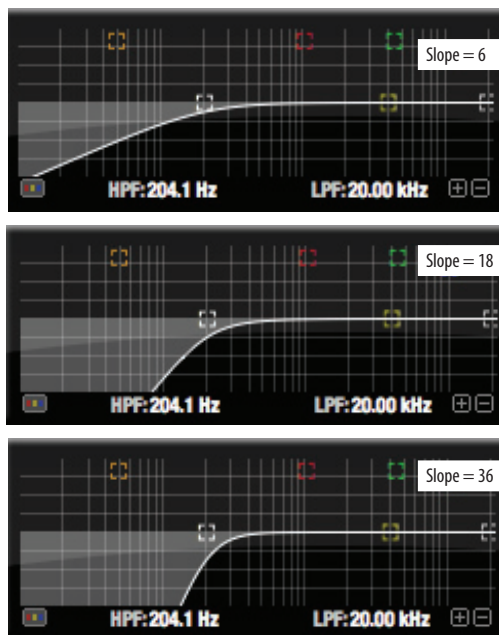


Figure 9-20: The low pass filter with three example slope settings.

The Dynamics tab

The Dynamics tab (Figure 9-21) displays the Dynamics processing settings for the input or output channel that currently has the focus. Click any focus button in the Inputs or Outputs tab to view the Dynamics tab settings for the channel.

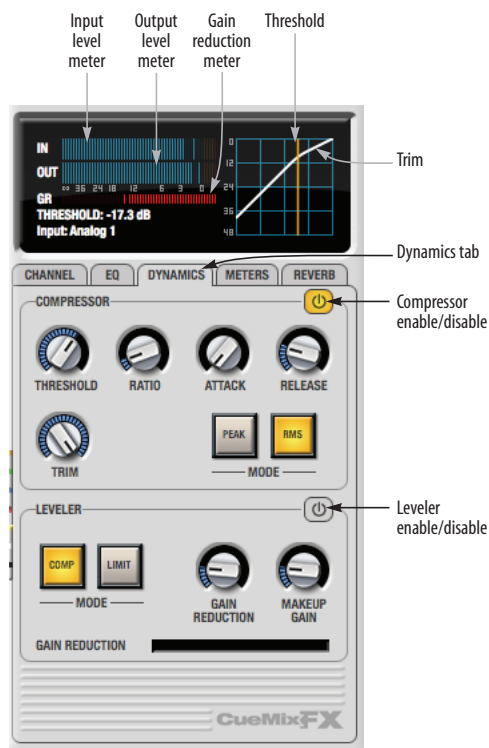


Figure 9-21: The Dynamics tab.

Enabling Dynamics

Each input and output channel has a global *Dynamics enable/disable* button (Figure 9-3 and Figure 9-6). This button enables or disables all dynamics processing for the channel. In addition, the Dynamics tab has two different dynamics processors, the Compressor and Leveler, which can be individually enabled or disabled (Figure 9-21) for the channel.

Compressor

The *Compressor* (Figure 9-21) lowers the level of the input when it is above the threshold. The amount of attenuation is determined by the *Ratio* and the input level. If the input is 6 dB above the *Threshold* and the Ratio is 3:1, then the output will be 2 dB above the Threshold. When the input level goes above the threshold, the attenuation is added gradually to reduce distortion. The rate at which the attenuation is added is determined by the *Attack* parameter. Likewise, when the input level falls below the Threshold, the attenuation is removed gradually. The rate at which the attenuation is removed is determined by the *Release* parameter. Long Release times may cause the audio to drop out briefly when a soft passage follows a loud passage. Short Release times may cause the attenuation to pump when the average input level quickly fluctuates above and below the Threshold.

These sorts of issues can be addressed by applying the Leveler instead.

Graphic adjustment of the Threshold

The Threshold can be adjusted by turning the Threshold knob or by dragging the Threshold line directly in the compressor graph (Figure 9-21).

Input level meter

The *Input Level* meter (Figure 9-21) shows the level of the input signal before it enters the compressor. It shows either the peak level or the RMS level, depending on which mode is currently chosen.

Gain reduction (GR) meter

The Gain reduction (*GR*) level meter (Figure 9-21) displays the current amount of attenuation applied by the compressor.

Output level

The *Output Level* meter (Figure 9-21) displays the peaks of the output signal. Trim is applied before the Output Level meter.

Peak/RMS modes

In *RMS mode* the compressor uses RMS values (a computational method for determining overall loudness) to measure the input level. In *Peak mode*, the compressor uses signal peaks to determine the input level. RMS mode will let peaks through because the detector sidechain is only looking at the average signal level. Peak mode will react to brief peaks. Peak mode is generally used for drums, percussion and other source material with strong transients, while RMS mode is mostly used for everything else.

The input meters show either the peak level or the RMS level, depending on the mode.

Leveler

The Leveler™ (Figure 9-21) provides an accurate model of the legendary Teletronix™ LA-2A® optical compressor, known for its unique and highly sought-after Automatic Gain Control (AGC) characteristics. The 896mk3 Leveler faithfully models the LA-2A using the on-board DSP with 32-bit floating point precision.

A model of an optical compressor

The simplest description of an optical leveling amplifier device is a light shining on a photoresistor. The intensity of the light source is proportional to the audio signal, and the resistance of the photoresistor is in turn inversely proportional to the intensity of the light. Photoresistors respond quite quickly to increases in light intensity, yet return to their dark resistance very slowly. Thus, incorporation of the photoresistor into an attenuator followed by an amplifier which provides make-up gain produces a signal which maintains a constant overall loudness.

Automatic gain control using light

The the Automatic Gain Control (AGC) circuit of the LA-2A uses a vintage opto-coupler known by its model number (T4). The T4 contains an electroluminescent panel (ELP) and photoresistor

mounted so that the emission of the panel modulates the resistance. An ELP consists of a thin layer of phosphorescent material sandwiched between two insulated electrodes to form a capacitor. Making one of the electrodes transparent allows the light to escape. These devices are essentially glow-in-the-dark paint on a piece of foil covered by metalized glass or plastic, and are the same devices used in low-power night lights. Unfortunately, these devices need high voltages to operate, and are best driven by tube circuits which can supply voltage swings of several hundred volts.

Response characteristics

Once the light has faded away, the photoresistor then decays back to its dark state. The shape of the decay curve varies depending on how bright the light was, and how long the light lasted. A general rule of thumb is that the louder the program, the slower the release. Typically, the release can take up to and over one minute. One thing to keep in mind when using these types of devices is that the typical concepts of compression ratio, attack, release, and threshold do not apply. The light intensity is determined by the highly non-linear interactions of the input signal, AGC circuit, and ELP, and thus exhibit a strong program dependence that is impossible to describe without the mind-numbing mathematics of statistical mechanics. The actual results, however, can be almost mystical: even when you feed the same material (a loop perhaps) through the Leveler twice, you'll often see a new response the second time through a loop, complete with unique attack times, release times and compression ratios. Furthermore, two different input signals with the same RMS levels may be leveled in a drastically different manner.

It is precisely this self-adjusting behavior that makes optical compressors the tool of choice for smoothing out vocals, bass guitar and full-program mixes without destroying perceived dynamics.

Compressor/Limit buttons

The *Comp* and *Limit* buttons (Figure 9-21) model the original LA-2A Limit/Compress mode switch. The effect is very subtle, with the Limit option behaving only slightly more like a limiter than a compressor. The switch increases the level of the input to the AGC model and runs the attenuator at a slightly lower level. The Leveler then responds more strongly to transients, but otherwise still behaves like a leveling amplifier.

Gain Reduction

Gain Reduction (Figure 9-21) sets the strength of the signal sent to the AGC model.

Makeup Gain

Makeup gain (Figure 9-21) amplifies the output signal to make up for gain reduction.

Enabling or disabling the Leveler

The Leveler models the LA-2A so closely, it also models the time it takes for an actual LA-2A to “warm up” after it is turned on. Therefore, when you enable the Leveler, give it a moment to “settle” before you begin processing signals with it.

The Meters tab

The *Meters* tab (Figure 9-22) serves as a comprehensive meter bridge for all inputs, outputs and mix busses in the 896mk3. This tab gives you a “bird’s-eye” view of all signal activity in the 896mk3; it is ideal for confirming your signal routing programming and for troubleshooting.

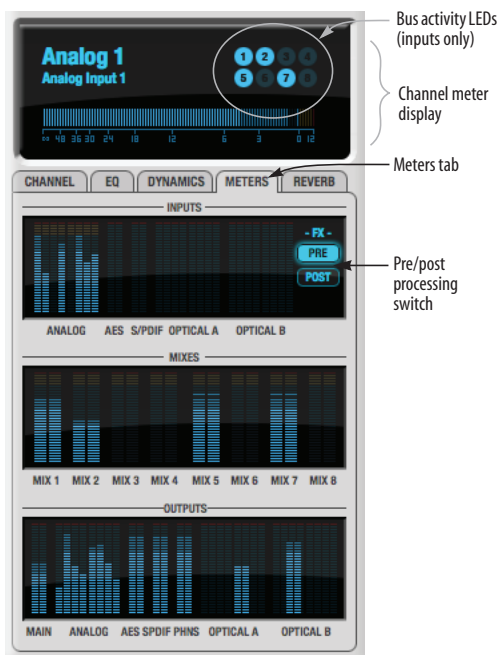


Figure 9-22: The Meters tab.

Channel meter display

The *channel meter display* (Figure 9-22) provides a long-throw meter for the input or output that currently has the focus in the Input/Output tabs.

Bus activity LEDs (inputs only)

The *Bus activity LEDs* (Figure 9-22) are present only for inputs. See “Input meter and bus activity LEDs” on page 83.

Pre/post processing switch

The *pre/post processing* switch (Figure 9-22) affects all input meters (and the meter in the channel meter display above the tab, if this area is

displaying an input meter). Click *Pre* to view levels *before* any input channel processing besides trim; click *Post* to view levels *after* all channel processing (EQ, compression, M/S decoding, L/R swap, etc.)

The Reverb tab

The *Reverb* tab (Figure 9-23) provides access to the 896mk3's single, global reverb processor, which provides high-fidelity reverberation and graphic control over its parameters.



Figure 9-23: The Reverb tab.

Enabling reverb

Use the enable/disable button (Figure 9-23) to turn the reverb processor on or off. Since reverb uses considerable DSP resources, it is best to leave it off when you are not using it.

Routing inputs, busses and outputs to the reverb processor

The reverb processor is a single, independent unit that provides stereo reverb. You can route multiple signals to it from various points (sends) in the CueMix FX mixer, but all incoming signals to the reverb processor are merged and processed together. The resulting stereo output from the reverb can then be inserted into a mix bus or output using stereo returns.

Reverb sends

The following signals can be sent to the reverb processor via their corresponding sends (discussed earlier in this chapter):

- Mono or stereo inputs (Figure 9-3 on page 77)
- Mix bus output (Figure 9-2 on page 75)
- Outputs (Figure 9-6 on page 80)

Reverb returns

The stereo output from the reverb processor can be sent to the following destinations via their corresponding returns (discussed earlier in this chapter):

- Mix bus outputs
- Outputs
- The computer (via the Reverb Return bus)

Split point

The *Split Point* (Figure 9-23) prevents feedback loops that would be caused by a signal being sent to the reverb processor and then returned to the same signal path.

Mix

When the Split Point is set to *Mix*, the returns in the Mix bus tab become active and the sends in the Output tab gray out. This allows you to send from inputs and mixes and return to mixes and outputs.

Output

When the Split Point is set to *Output*, the sends in Output tab become active and the returns in the Mix bus tab gray out. This allows you to send from inputs, mixes and outputs and return to outputs.

Primary controls

The Primary Controls section (Figure 9-23) in the Reverb tab provides the following basic parameters for programming the reverb.

Reverb Time

Reverb time determines the length of decay, or *tail*, of the reverb. This is a global setting for the reverb processor. You can further refine the tails by independently setting the reverb time of three separate frequency bands, as discussed below in the Reverb Design section.

PreDelay

PreDelay is the amount of time before you hear the very first reflections. If you are in a large room, it takes a while before the first reflections return. PreDelay is useful for clarifying the original sound. For example, with vocals, the reflections won't start until after the initial sound of a word has been sung.

Shelf Filter

The *Shelf Filter* is a low-pass filter that controls the high frequency characteristics of the overall effect. *Frequency* sets the cutoff frequency for the filter and *Cut* sets the amount of signal attenuation applied by the filter.

Early reflections

Initial reflections give a space its unique sound. The shape of the room, the angles of the walls, even furniture in the room will produce a series of Initial Reflections. Think of the early reflections and room type as the “flavor” of the reverb. You can choose between several types of rooms. These are acoustic models for simulating these different

types of spaces. The *Size* and *Level* parameters let you control the size of the room and the strength of the initial reflections.

☞ Here's a tip: try using initial reflections without any subsequent reverb (turn the reverb time down as far as it will go). You'll hear interesting and unusual effects.

Reverb design

The Reverb Design section allows you to independently control the reverb time for three separate frequency bands (*Low*, *Mid* and *High*) with adjustable cross-over points between them (*Low* and *High*). The *reverb time* for each band is specified in percent of the overall reverb time in the Primary Controls section at the top of the tab.

You can edit these parameters graphically by dragging the handles in the graphic display (Figure 9-23).

Width does what its name implies: if you turn this control all the way up, the result is maximum stereo imaging. A position of 12 o'clock produces essentially a mono image. Turning the control all the way down completely swaps the stereo image.

THE MONITOR GROUP

The *monitor group* is a set of 896mk3 outputs that can be controlled with the master Monitor Level knob in upper right corner of the CueMix FX mixer window (Figure 9-24), as well as the MASTER VOL knob on the 896mk3 front panel.

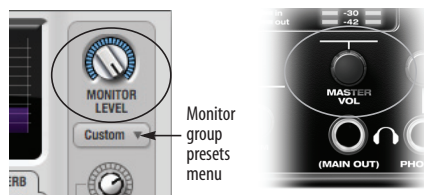


Figure 9-24: Monitor group volume control from CueMix FX and the front panel MASTER VOL knob.

Assigning outputs to the monitor group

Any combination of outputs can be assigned to the monitor group. To include an output pair in the monitor group, click its Monitor button in the Outputs tab (Figure 9-6 on page 80).

Monitor group presets menu

The *monitor group presets menu* (Figure 9-24) provides several presets for commonly used monitor groups:

Monitor group preset	Output assignment
Main Outs	Main Out 1-2
Stereo	Analog Out 1-2
Quad	Analog Out 1-4
5.1	Analog Out 1-6
7.1	Analog Out 1-8

If you program your own monitor output group, the presets menu displays the word *Custom*.

Monitor group meters

The monitor group meters (Figure 9-1 on page 73) show levels for any/all output pairs that are currently included in the monitor group. The width of the meters scales proportionally so that all current monitor group outputs will fit within the prescribed space for the meters. If many outputs are included, then the meters will look fairly thin, but they will all be included in the meter.

DSP METER

The DSP meter (Figure 9-1) shows how much of the available DSP processing power is currently being used by the 896mk3 for effects processing. DSP resources are allocated in channel order from the first input to the last output. If there aren't enough DSP resources for all effects to be enabled on a channel, none of them are allocated on that channel or any following channel.

EQ on a stereo channel requires approximately twice the DSP resources as the same EQ on a mono channel. The Compressor (2.5 x 1 EQ band) and Leveler (4 x 1 EQ band) require about the same DSP resources for a mono or stereo channel.

SOLO LIGHT

The Solo light (Figure 9-1) illuminates when any input in the current (active) mix bus is soloed (even if it is currently scrolled off-screen).

TALKBACK AND LISTENBACK

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

Hardware setup

Figure 9-25 below shows a typical hardware setup for Talkback and Listenback. For Talkback, set up a dedicated mic in your control room and connect it to a mic input on your MOTU audio interface. For Listenback, set up a dedicated listenback mic in the live room for the musicians and connect it to another mic input (or just use one of the mics you are recording from). For talkback output, set up a headphone distribution amp or set of speakers in the live room, and connect it to any 896mk3 output, as demonstrated below in Figure 9-25.

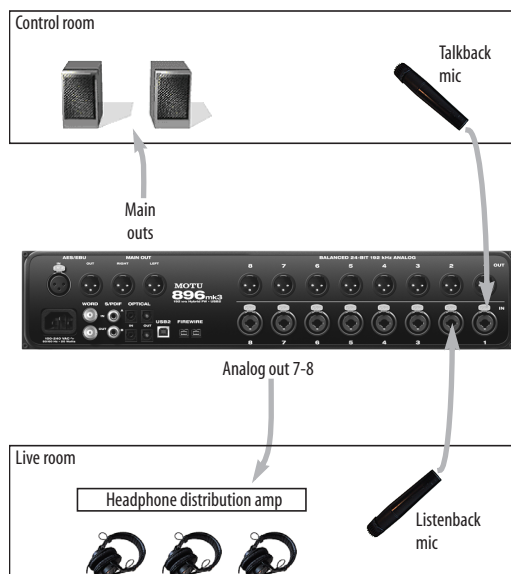


Figure 9-25: Typical hardware setup for Talkback and Listenback.

Talkback / Listenback Mic Input

To configure the talkback mic in CueMix FX, go to the Inputs tab (Figure 9-3 on page 77) and click the Focus button for the input that the talkback mic is connected to. Click the Channel tab (Figure 9-8 on page 82) and enable the *Talk* button.

Repeat this procedure for the Listenback mic, except click the *Listen* button in the Channel tab.

Talk / Listen output

To configure the talkback and listen back outputs, go to the Outputs tab (Figure 9-6 on page 80) and enable the *Talk* button for any output pair on which you'd like to hear the talkback mic. Similarly, enable the *Listen* button for any output pair on which you'd like to hear the listenback mic.

Talkback / Listenback Monitor Dim

Use the knobs next to the Talk and Listen buttons (Figure 9-26) to determine the amount of attenuation you would like to apply to all other audio signals (besides the talkback/listenback signal) when Talkback and/or Listenback is

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



Figure 9-26: The Talkback/Listenback controls.

Engaging/disengaging Talkback and Listenback

To engage Talk back or Listenback, press on the Talk or Listen buttons (Figure 9-26) and then release to disengage. Talkback and/or Listenback is engaged for as long as you hold down the mouse button. Alt-click to make the buttons “sticky” (stay engaged until you click them again — so you don’t have to hold down the mouse). Or use the Talkback menu items.

If you would like to engage both Talkback and Listenback at the same time, enable the *Link* button (Figure 9-26).

Controlling Talkback and Listenback volume

To control the volume of the Talkback and/or Listenback mics, adjust their input trim in CueMix FX.

SHORTCUTS

Hold down the following general modifier keys as shortcuts:

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

Hold down the following modifier keys as shortcuts for the EQ tab and controls:

Shortcut	Result
Shift click	Applies EQ button change to all input or outputs.
Alt-click	Applies EQ enable button changes to all bands in that input or output.
Shift-Alt-click	Applies EQ enable button changes to all bands and all inputs or outputs.

FILE MENU

Saving and loading hardware presets

The 896mk3 can store up to 16 presets in its on-board memory. A preset includes all CueMix FX settings for all mix busses, but it excludes global settings like clock source and sample rate.

The Load Hardware Preset and Save Hardware Preset commands in the CueMix FX file menu let you name, save and load presets in the 896mk3.

Peak/hold time

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

Mix1 Return Includes Computer

The *Mix1 return includes computer* File menu item applies to other MOTU interfaces products and has no effect on the 896mk3.

Hardware follows CueMix Stereo Settings

This File menu item applies to other MOTU interfaces products and has no effect on the 896mk3.

EDIT MENU

Undo/Redo

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

Copying & pasting (duplicating) entire mixes

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

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

Clear Peaks

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

DEVICES MENU

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

Audio analysis tools

Below each device are its signal analysis tools. Choose one to open its window. For details on these features, see the following sections.

FFT and Spectrogram display..... 99

Oscilloscope 102

X-Y Plot 108

Phase Analysis 111

Tuner 115

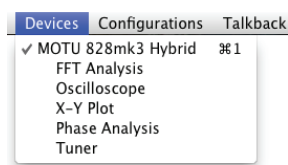


Figure 9-27: Opening the signal analysis windows.

Choosing channels for audio analysis

The audio analysis tools follow the currently focused audio input or output. (See “Channel focus and settings” on page 74.) If you focus a mono channel (e.g. Analog 3), its corresponding stereo pair will be displayed (Analog 3–4).

Scoping host software audio output

If you want to scope audio output from your host software, send your host’s output to an 896mk3 output pair, and then set the focus on that output pair.

FFT AND SPECTROGRAM DISPLAY

FFT and spectrogram information can be displayed in the Filter response display section in the EQ tab (Figure 9-10 on page 84) or as a separate window (Figure 9-29 on page 100) opened from the Devices menu (Figure 9-27).

Filter display options

The *Filter display options* menu (Figure 9-28) provides several options for the EQ filter display:

Menu option	What it does
Show no analysis	Turns off both the FFT and Spectrogram in the Filter display.
Show FFT	Shows/hides a real time FFT analysis of the current signal being EQ’d, post EQ filter.
Show Spectrogram	Shows/hides a real-time spectrogram “waterfall” in the background of the filter display, post EQ filter.
Show Band Response	Shows/hides the colored area below EQ filter points.
Show/Hide Full Window Analysis	Shows/hides the enlarged filter display in the CueMix FX window.

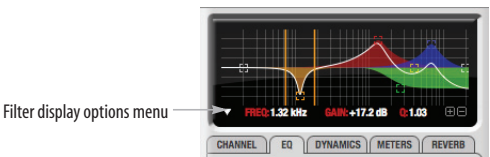


Figure 9-28: Filter Display options menu

FFT display

Choose *Show FFT* from the *Filter display options* menu (Figure 9-10) to superimpose a real-time Fast Fourier Transform (FFT) frequency measurement curve over the EQ filter display, as demonstrated in Figure 9-30:

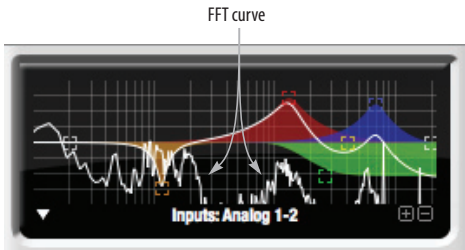


Figure 9-30: FFT display.

The FFT curve is post-filter. Therefore, the FFT shows the results of the EQ filter(s) being applied. Use the global EQ button for the input or output channel (Figure 9-3 and Figure 9-6, respectively) to toggle between the EQ'd and non-EQ'd FFT display for an A/B comparison.

Spectrogram

Choose *Show Spectrogram* from the *Filter display options* menu (Figure 9-10) to superimpose a real-time spectrogram “waterfall” display in the background of the EQ filter display, as demonstrated in Figure 9-31:

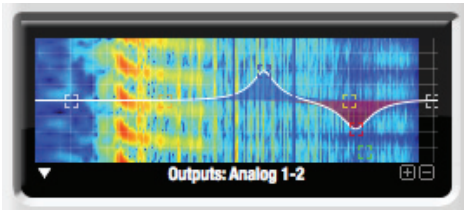


Figure 9-31: FFT display.

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



Figure 9-32: Spectrogram color-to-amplitude spectrum.

Opening the FFT Analysis window

Choose *FFT Analysis* from the *Devices* menu to open a new window with the filter EQ display for detailed inspection and adjustment of the EQ filter, as shown Figure 9-29.

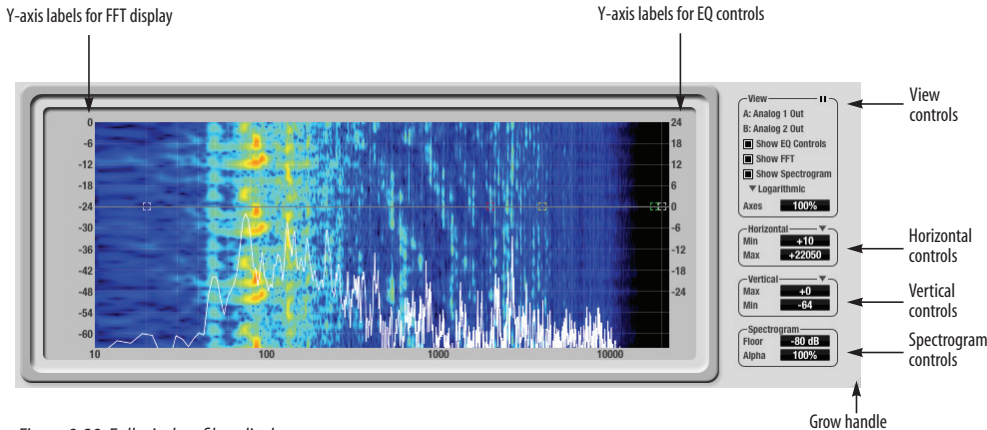


Figure 9-29: Full window filter display.

View controls

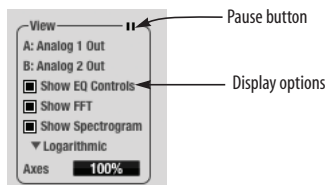


Figure 9-33: View controls

You can show and hide the FFT display, spectrogram or EQ band response curves as desired using the *Display Options* menu options (Figure 9-29). These settings are independent of the small graph display options (Figure 9-10), so you have the flexibility to display different combinations in each graph.

☞ “Show EQ Controls” will be available only if the focused pair are a stereo input pair or stereo output pair.

Logarithmic or Linear X-Axis Scale

The x-axis defaults to a logarithmic scale, but can be changed to a linear scale if desired. In the View controls (Figure 9-33), click *Logarithmic* to access the x-axis scale options menu. With a linear scale selected, frequency is constant, but the width of each octave along the x-axis is different. With a logarithmic scale selected, octaves are displayed with a constant width, but frequency is displayed logarithmically within each octave.

Axes display

The Axes control (Figure 9-33) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

Pausing the display

The Pause button in the upper right corner of the View section (Figure 9-33) allows you to freeze the display at any time. To resume, click the button again.

The info box

When any EQ filter point is selected or dragged in the full window graph, the info box is shown next to the point in the full graph display (Figure 9-34).

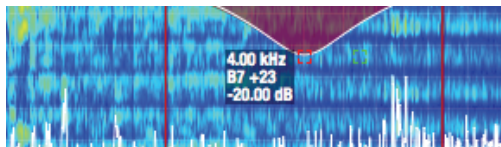


Figure 9-34: The Info Box.

The info box includes the industry standard scientific note (pitch) name when the control point is located at a frequency that resides within a prescribed note range, where C4 is middle C. The note number is accompanied by the number of cents (± 50) above or below the exact frequency for the note. If the control point is dragged outside the note range, only the frequency is shown.

Horizontal controls (frequency axis)

The *Horizontal* controls (Figure 9-35) configure the value range of the x-axis (frequency). Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the Horizontal control menu (Figure 9-35).

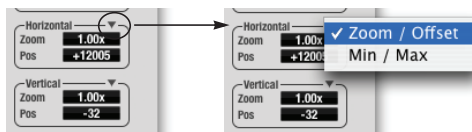


Figure 9-35: Horizontal control menu

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

one half of the entire frequency range is displayed. *Pos* determines which frequency is displayed at the center of the graph.

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

Vertical controls (amplitude axis)

The *Vertical* controls (Figure 9-29) operate similarly to the Horizontal controls, except that they configure the y-axis (amplitude). In *Zoom/Offset* mode, *Zoom* sets the display zoom from 1x to 100x, and *Pos* sets the center amplitude of the graph. In *Min/Max* mode, *Min* and *Max* set the smallest and largest displayed amplitude.

Spectrogram controls

The *Floor* control (Figure 9-29) sets the amplitude threshold for the spectrogram display, from -144 dB up to 0 dB.

The *Alpha* control (Figure 9-29) sets the opacity of the spectrogram information displayed in the graph, from 100% (fully visible) to 0% (hidden).

OSCILLOSCOPE

The Oscilloscope (Figure 9-36) graphs the amplitude of an audio signal over time.

Amplitude is displayed on the y-axis and time is displayed on the x-axis. A thick white vertical line marks where time equals zero; a thick white horizontal line marks where amplitude equals zero (Figure 9-36, below).

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

Opening the oscilloscope

Each 896mk3 has its own oscilloscope. To open an oscilloscope, choose the *Oscilloscope* item from the Devices menu under the desired interface.

Choosing a channel to display

The oscilloscope follows the currently focused audio input or output. If you focus a mono channel (e.g. Analog 3), its corresponding stereo pair will be displayed (Analog 3–4).

View controls

The View controls (Figure 9-37) provide several options for the oscilloscope display.

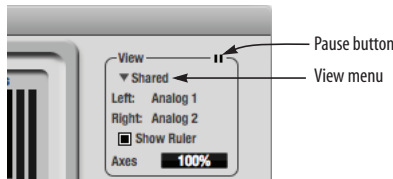


Figure 9-37: View controls

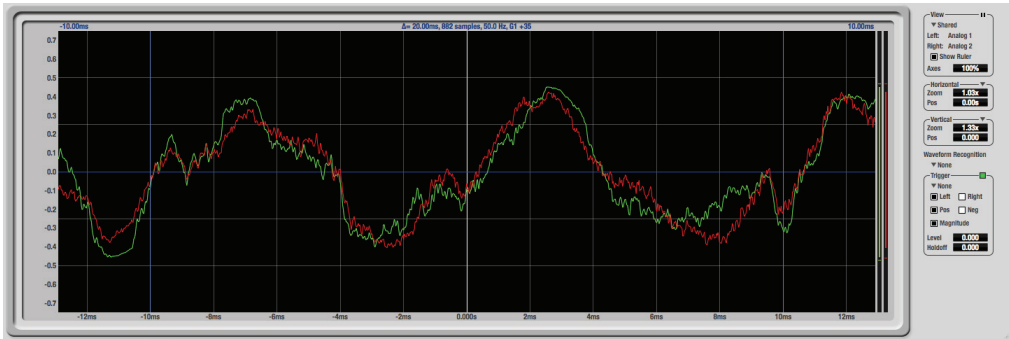


Figure 9-36: Oscilloscope

View menu

The View menu (Figure 9-37) lets you choose how you wish to display the audio channel(s) being displayed.

View menu settingWhat it displays

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

Display options

The Axes control (Figure 9-37) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden). The Show Ruler option toggles the measurement items (see “Measurement information” on page 105).

Pausing the display

The Pause button in the upper right corner of the View section (Figure 9-37) allows you to freeze the display at any time. To resume, click the button again. The level meters will remain active while the display is paused.

Horizontal controls (time axis)

The Horizontal controls (Figure 9-38) configure the value range of the x-axis (time). Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the Horizontal control menu (Figure 9-38).

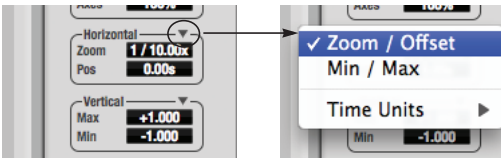


Figure 9-38: Horizontal control menu

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

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

Time Units

The *Time Units* sub-menu (Figure 9-38) provides the option to view the X axis in Seconds or Samples.

Vertical controls (amplitude axis)

The Vertical controls (Figure 9-38) operate similarly to the Horizontal controls, except that they configure the y-axis (amplitude).

In *Zoom/Offset* mode, *Zoom* sets the display zoom from 1/2 to 100x, and *Offset* moves the line marking amplitude equals zero line up or down.

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

Waveform Recognition

The Waveform Recognition option searches through new audio data looking for a waveform which most resembles that which was previously displayed. The region where this takes place is a small window around the line marking time equals

zero, denoted by the extra vertical graph lines surrounding it. There are two kinds of waveform recognition available: Type I and Type II.

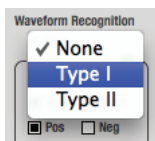


Figure 9-39: Waveform Recognition menu

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

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

Trigger

When the *Trigger* (Figure 9-40) is not enabled (the Trigger menu is set to *None*), the graph updates based on time: after every *n* samples of the monitored audio signal, the most recent samples are displayed. When the Trigger is enabled (set to any mode other than *None*), the graph updates in response to specific conditions in the signal. The Trigger section defines that criteria and how the graph will display the events that match.

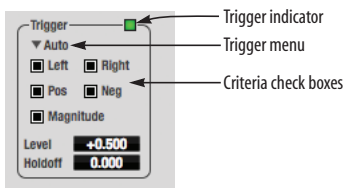


Figure 9-40: Trigger settings

Criteria

The criteria checkboxes (Figure 9-40) determine the conditions that the trigger is looking for and where it will look for them.

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

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

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

Enabling the *Magnitude* checkbox tells the trigger to look for both positive and negative Level values, regardless of whether the Level value is positive or negative. For example, if Level is set to +0.500 and *Magnitude* is enabled, the trigger will look for both +0.500 and -0.500. You will see a second blue line appear in the display when *Magnitude* is enabled to denote the second value.

Holdoff

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

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

Trigger modes

The Trigger menu (Figure 9-40 on page 104) provides four modes:

Trigger mode	What it does
None	The Trigger is not active; this is the default mode. The incoming audio signal will be displayed continuously as audio is received.
Auto	The display is always updating, but when the condition is met, the trigger event will be displayed centered around the line marking time equals zero.
Normal	The display updates only when the condition is met; the last trace will be displayed until the next matching event is found.
Single Sweep	Similar to Normal mode, but the last trace will be displayed until you manually arm the trigger by clicking the Trigger indicator (Figure 9-40 on page 104) or by pressing the spacebar.

Trigger indicator

The Trigger indicator (Figure 9-40 on page 104) displays the state of the trigger, and also provides a way to manually interact with it. The Trigger indicator always displays one of three colors:

Color	Status
Green	When the current Trigger criteria has been met (including when the Trigger mode is <i>None</i>).
Yellow	When the Trigger is armed, but has not yet found an event which matches its criteria. Yellow can also indicate that the graph has been manually paused using the Pause button in the View section (see “Pausing the display” on page 103).
Red	When the Trigger is being held off, either because the Trigger mode is set to Single Sweep or the Holdoff time is not set to zero.

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

Measurement information

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

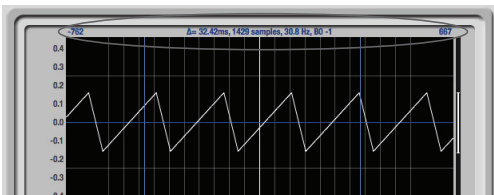


Figure 9-41: Measurement information

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

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

Ideas for using the Oscilloscope

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

Analyzing and comparing harmonic content

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

Viewing transients such as drum hits

If you loop a snare hit or other similar transient audio clip and feed it through the oscilloscope, you can more or less “freeze” the transient waveform in the oscilloscope frame. This can be useful, for example, for viewing the results of real-time compression that you are applying with an effects plug-in, as demonstrated in Figure 9-42. In this example, a snare hit is being compressed by Digital Performer’s Dynamics plug-in. As you make adjustments to the compression plug-in’s settings, you can see the transient waveform change the next time the Oscilloscope triggers. For compression, this can be particularly useful for balancing the effect of the attack on the transient, relative to the decay portion of the waveform. Conversely, you can see the effect of the threshold setting directly on the decay portion, relative to the attack. In effect, you can see as well as hear the results of your compression adjustments.

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

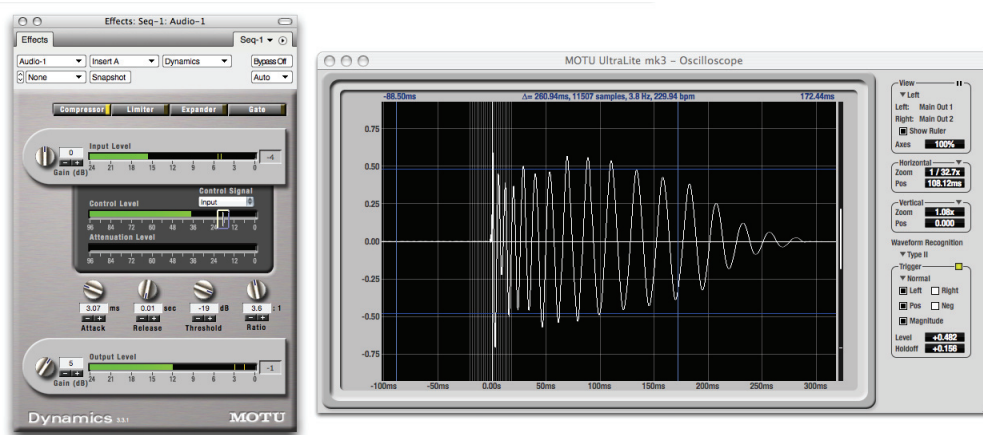


Figure 9-42: Viewing transients in the Oscilloscope

transient is settled in the display and fairly stable, you may need to adjust the horizontal position to center it in the display. These settings are depicted in the example in Figure 9-42.

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

Clip detection

You can use the Oscilloscope to detect clipping in a digital audio signal. To do so, enable all criteria (Figure 9-40 on page 104), choose *Single Sweep* from the trigger menu (Figure 9-40), set the level to 0.999 and click the trigger indicator (Figure 9-40) to arm it (yellow). As soon as the signal clips, the trigger indicator will turn red, and the display will show the offending clip at the line marking time equals zero.

Viewing timing pulses

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

Building synthesizer patches

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

Another example is waveform polarity. If you are combining several raw waveforms, polarity is a critical, yet not always obvious, factor in

determining the resulting sound. You can use the Oscilloscope to easily view and compare polarities to see if they are inverted from one another or not. The *Add* and *Subtract L - R* View menu settings are particularly useful here.

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

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

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

Monitoring control voltage output from Volta

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

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

X-Y PLOT

The *X-Y Plot* window (Figure 9-43) graphs the amplitude of a stereo audio signal on a two-dimensional grid.

For each unit of time (i.e., each sample), the amplitude of the left channel is displayed on the x-axis and the amplitude of the right channel is displayed on the y-axis. A thick white vertical line marks where left channel amplitude equals zero; a thick white horizontal line marks where right channel amplitude equals zero (Figure 9-43, below). There are also thick white diagonal lines for $y = x$ and $y = -x$.

Metering

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

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

Opening the X-Y Plot

Each 896mk3 interface has its own X-Y Plot window. Choose the *X-Y Plot* item from the Devices menu under the desired interface.

Choosing a channel pair to display

The X-Y Plot follows the currently focused audio input or output. If you focus a mono channel (e.g. Analog 3), its corresponding stereo pair will be displayed (Analog 3–4).

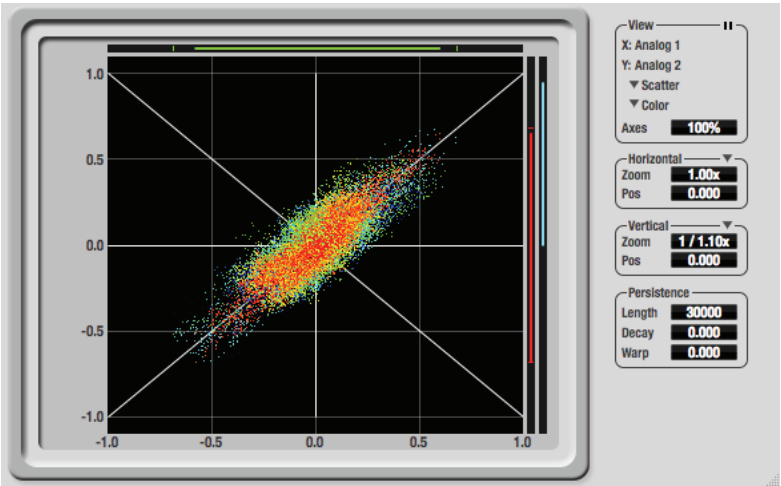


Figure 9-43: X-Y Plot

View controls

The View controls (Figure 9-44) provide several options for the X-Y Plot display.

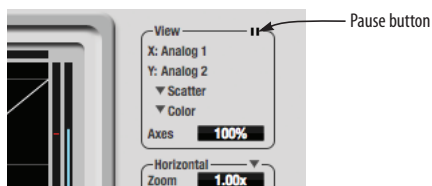


Figure 9-44: View controls

Pausing the display

The Pause button in the upper right corner of the View section (Figure 9-44) allows you to freeze the display at any time. To resume, click the button again. The level meters will remain active while the display is paused.

Line/Scatter

Choose either *Line* or *Scatter* from the menu in the View section (Figure 9-44) to plot each point (sample) as either a single pixel or as a continuous line that connects each plot point to the next, as shown below in Figure 9-45.

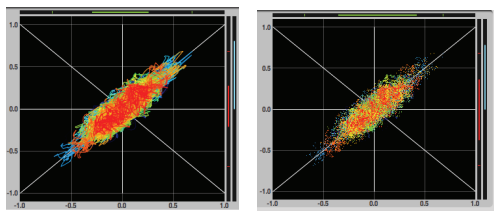


Figure 9-45: The same X-Y Plot displayed in Line versus Scatter mode.

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

Color/Grayscale

In *Color* mode (Figure 9-44) the most recently displayed audio data is shown in red, which fades to yellow, green and then finally blue, before disappearing. In *Grayscale* mode, data is first

shown in white and then fades to gray. To adjust the scale of this color/brightness change, see “Decay” on page 110.

Axes

The *Axes* control (Figure 9-44) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

Horizontal and vertical controls

The *Horizontal* and *Vertical* controls (Figure 9-46) configure the value range of the x-axis (left channel amplitude), and y-axis (right channel amplitude), respectively. Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the menu shown in Figure 9-46.

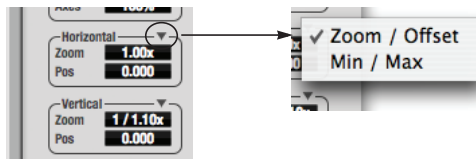


Figure 9-46: Setting the Horizontal or Vertical control modes.

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

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

Persistence

The *Persistence* controls (Figure 9-47) affect the appearance of data from when it is first displayed until it disappears from the grid.

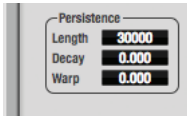


Figure 9-47: The Persistence controls.

Length

Length (Figure 9-47) sets the number of recent samples to show on the plot. For example, when Length is set to 10,000, the 10,000 most recent samples are shown.

Decay

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

Decay (Figure 9-47 on page 110) determines the brightness or hue of the minimum value. When Decay is zero, the oldest sample is black. When Decay is +1.000, the oldest sample is fully opaque (in Grayscale mode) or red (in Color mode).

Warp

Warp (Figure 9-47) determines the position of data points after they are first drawn. When warp is zero, data points remain in the same position. When warp is positive, they contract towards the origin (center of the grid). When warp is negative, they expand away from the origin. The further the warp value is from zero, the greater the effect.

Using the X-Y Plot

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

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

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

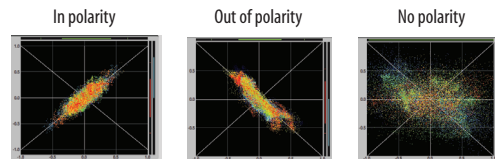


Figure 9-48: Checking polarity in a stereo signal with the X-Y Plot.

PHASE ANALYSIS

The *Phase Analysis* window (Figure 9-49 on page 111) graphs frequency versus phase difference versus amplitude of a stereo signal on either rectangular or polar coordinates.

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

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

Correlation Meter

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

Opening the Phase Analysis

Each 896mk3 interface has its own Phase Analysis window. Choose the *Phase Analysis* item from the Devices menu under the desired interface.

Choosing a channel pair to display

The Phase Analysis window follows the currently focused audio input or output. If you focus a mono channel (e.g. Analog 3), its corresponding stereo pair will be displayed (Analog 3–4).

View controls

The View controls (Figure 9-50) provide several options for the Phase Analysis display.

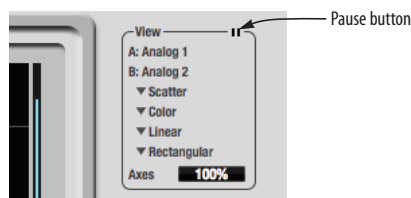


Figure 9-50: View controls

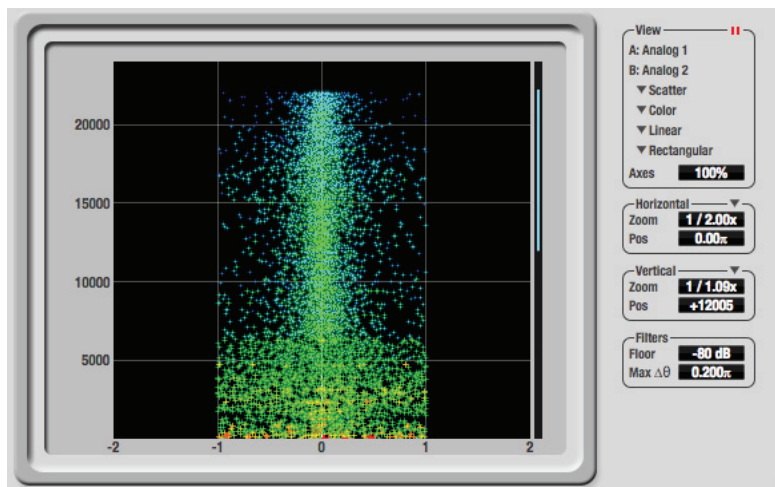


Figure 9-49: Phase Analysis

Pausing the display

The Pause button in the upper right corner of the View section (Figure 9-50) allows you to freeze the display at any time. To resume, click the button again. The correlation meter will remain active while the display is paused.

A/B (stereo audio channels)

The View section (Figure 9-50) displays the pair of input or output audio channels you are viewing. See “Choosing a channel pair to display” above.

Line/Scatter

Choose either *Line* or *Scatter* from the menu in the View section (Figure 9-50) to plot each data point as either a single pixel or as a continuous line that connects each frequency data point to the next, as shown below in Figure 9-45.

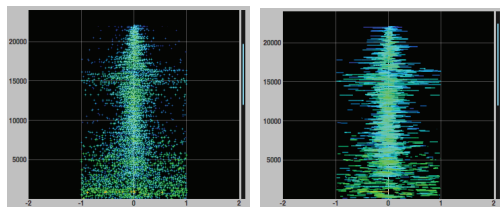


Figure 9-51: The same Phase Analysis displayed in Line versus Scatter mode.

Line mode is significantly more CPU intensive than Scatter. You can reduce Line mode CPU overhead for the Phase Analysis display by increasing the Floor filter and reducing the Max Delta Theta filters (see “Filters” on page 113).

Color/Grayscale

In *Color* mode (Figure 9-50) signal amplitude is indicated by color as follows: red is loud and blue is soft. In grayscale mode, white is loud and gray is soft.

Linear/Logarithmic

Choose either *Linear* or *Logarithmic* from the menu in the View section (Figure 9-50) to change the scale of the frequency axis. In rectangular coordinates, the vertical axis represents frequency,

and in polar coordinates, the radius from the center is frequency. With a linear scale, frequencies are spaced evenly; in a logarithmic scale, each octave is spaced evenly (frequencies are scaled logarithmically within each octave).

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

Rectangular/Polar

Choose either *Rectangular* or *Polar* from the menu in the View section (Figure 9-50) to control how audio is plotted on the Phase Analysis grid.

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

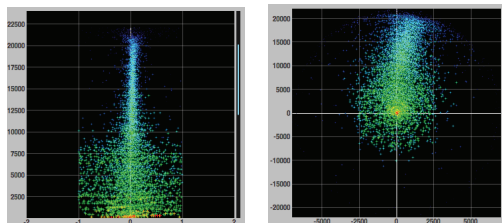


Figure 9-52: Rectangular versus Polar display (with a linear plot).

Above, Figure 9-52 shows Rectangular versus Polar display with a Linear plot. Below, Figure 9-53 shows the same displays (and the same data) with a Logarithmic plot:

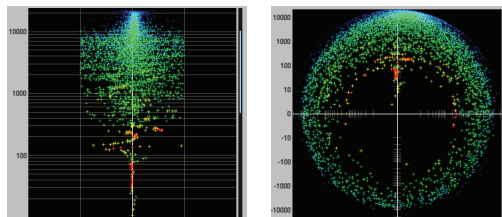


Figure 9-53: Rectangular versus Polar display with a logarithmic plot.

Axes

The *Axes* control (Figure 9-50) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

Horizontal and vertical controls

The *Horizontal* and *Vertical* controls (Figure 9-54) let you scale each axis of the grid and offset its zero point. Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the menu shown in Figure 9-54.

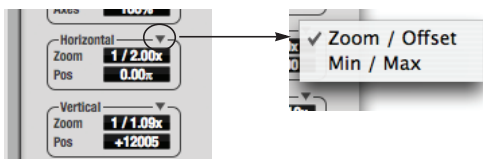


Figure 9-54: Setting the Horizontal or Vertical control modes.

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

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

Filters

The *Filters* section (Figure 9-55) lets you control the density of the Phase Analysis display.

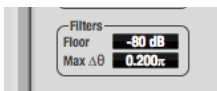


Figure 9-55: Filters

Floor

Floor (Figure 9-55) determines the amplitude threshold for the display. When the amplitude of both channels drops below this threshold, the signal is not shown.

Max delta theta

Max delta theta (Figure 9-55) only affects Line view (see “Line/Scatter” on page 112) and sets the maximum difference in frequency between plot points in the line plot. For two adjacent frequencies, if the distance (phase difference) between the two frequencies is greater than the *Max delta theta*, then the line is not drawn.

Using the Phase Analysis

In the polar display (top row of Figure 9-56 on page 114), stereo material that is predominantly phase-aligned (correlated) appears along the vertical axis, as demonstrated in the first column (*Perfectly in phase*) in Figure 9-56. If the vertical line tilts left or right, this indicates general differences in phase; the more the tilt (delta theta), the more the phase difference. If the vertical line points downwards in the polar display, this indicates that the stereo image is predominantly out of polarity, as demonstrated by the fourth column (*Inverted*) in Figure 9-56. Delays appear as spirals in the polar display.

The rectangular display (bottom row of Figure 9-56) also shows a predominantly phase-aligned stereo image along the vertical axis, and tilt (or left-right offset) from the center vertical axis represents differences in phase. If a signal is predominantly out of polarity, it appears along the $\theta = -1.0$ or $\theta = +1.0$ lines in the rectangular display, as demonstrated in the fourth column (*Inverted*) in Figure 9-56 on page 114.

Using Phase Analysis for multiple mic placement

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

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

Tuning PA systems

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

Summing to mono

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

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

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

Checking for phase issues in stereo tracks

You can use the Phase Analysis window to check the overall polarity of a stereo mix. Figure 9-57 is an example of a full stereo mix that has phase issues, as indicated by the majority of the signal's energy, which is predominantly skewed to the left side of the rectangular view (left) and spread along the -y axis in the polar view (right).

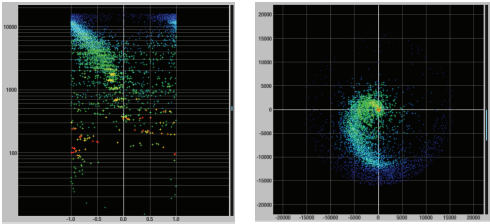


Figure 9-57: A stereo mix with phase issues.

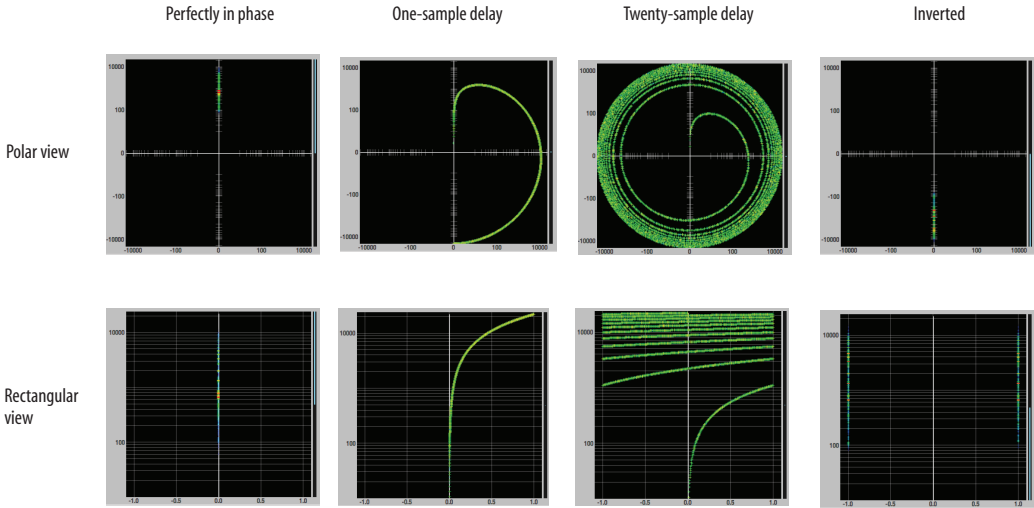


Figure 9-56: Two identical audio streams in the Phase Analysis.

TUNER



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

Opening the Tuner

Each 896mk3 interface has its own *Tuner* window. Choose the *Tuner* item from the *Devices* menu under the desired interface.

Choosing a channel pair to display

The *Tuner* window follows the currently focused audio input or output. If you focus a mono channel (e.g. Analog 3), its corresponding stereo pair will be displayed (Analog 3–4).

Choosing the channels to tune

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

Tuner controls

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

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

Meter: representation of the pitch difference between the detected note and the detected fundamental frequency. The horizontal position of the illuminated segments indicates how far the detected frequency is from the detected note. The number of illuminated segments indicates uncertainty or inharmonicity in the signal; a greater number of illuminated segments represents greater uncertainty. The color of the segments changes gradually from green (in tune) to yellow, orange, and red (progressively further out of tune).

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

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

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

Tuning stereo signals

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

CONFIGURATIONS MENU

A configuration is just like a hardware preset (a “snapshot” of all settings in CueMix FX and therefore the 896mk3 hardware itself), except that it can be created and managed using the CueMix FX software on your computer, completely independently of the 896mk3 hardware. The commands in the Configurations menu let you create, save, load, import, export and otherwise manage as many configurations as you wish.

Here is a summary of Configurations menu operations:

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

Modifying a configuration

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

to another configuration, choose *Save To*. To save as a new, separate configuration, choose *Create New*.

Saving a CueMix FX configuration as a hardware preset

To save a CueMix FX configuration as a hardware preset:

- 1 Choose the configuration from the Configurations menu to make it the current active configuration.
- 2 Choose *File menu > Save Hardware Preset*.
- 3 Type in a name, choose a preset slot and click OK.

Saving a hardware preset as a CueMix FX configuration

To save a hardware preset as a CueMix FX configuration:

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

TALKBACK MENU

Choose the commands in the Talkback menu to engage or disengage Talkback or Listenback.

PHONES MENU

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

CONTROL SURFACES MENU

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

Application follows control surface

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

Share surfaces with other applications

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

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

CueMix Control Surfaces

CueMix FX includes support for the following control surface products:

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

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

Enabled

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

Configure...

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

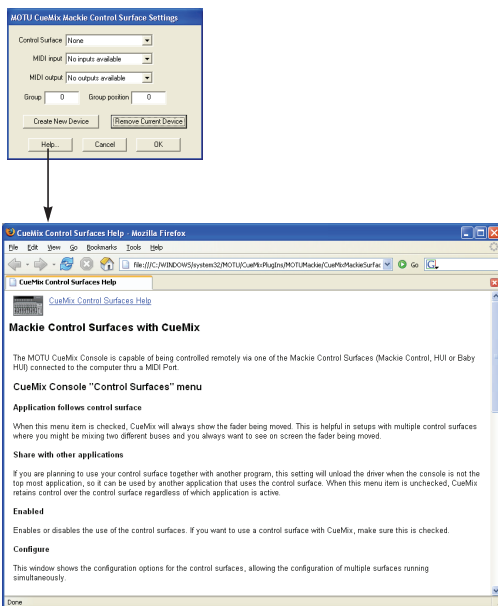


Figure 9-58: Refer to the extensive on-line help for details about configuring CueMix FX for operation with your control surface product.

CHAPTER 10 MOTU SMPTE Console

OVERVIEW

The 896mk3 can resolve directly to SMPTE time code via any analog input, without a separate synchronizer. The 896mk3 can also generate time code via its time code output. The 896mk3 provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy. Direct time code synchronization is supported by Cubase, Nuendo and other audio sequencer software that supports the ASIO2 sample-accurate sync protocol.

MOTU SMPTE Console	119
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Frame Rate	119
Reader section	120
Generator section	121
Setting up for SMPTE time code sync	121
Resolving Cubase or Nuendo to time code	121

MOTU SMPTE CONSOLE

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

CLOCK/ADDRESS

The *Clock/Address* menu (Figure 10-1) provides the same global *Master Clock Source* setting as in MOTU Audio Console (“Master Clock Source” on page 41), but it includes additional information: each setting shows both the clock and the address (time code or sample location), separated by a forward slash (/). To resolve the 896mk3 to SMPTE time code, choose the *SMPTE / SMPTE* setting in the Clock/Address menu. This means that the system will use SMPTE as the clock (time base) and SMPTE as the address.

FRAME RATE

This setting should be made to match the SMPTE time code frame rate of the time code that the system will be receiving. The 896mk3 can auto-

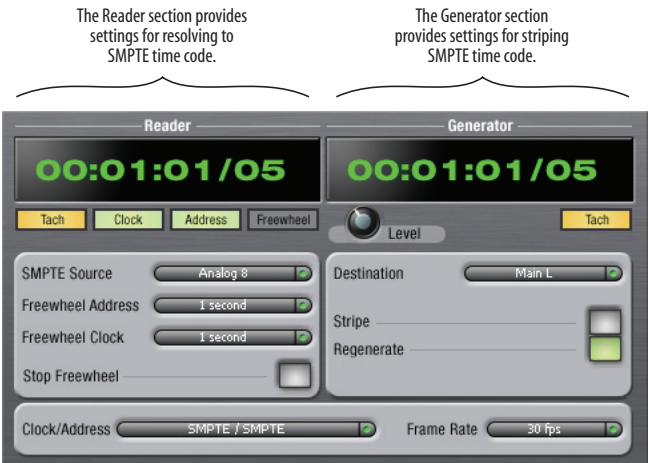


Figure 10-1: SMPTE Console gives you access to your 896mk3’s on-board SMPTE time code synchronization features.

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

READER SECTION

The Reader section (on the left-hand side of the window in Figure 10-1) provides settings for synchronizing the 896mk3 to SMPTE time code.

Status lights

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

Tach

The Tach light blinks once per second when the 896mk3 has successfully achieved lockup to SMPTE time code and SMPTE frame locations are being read.

Clock

The Clock light glows continuously when the 896mk3 has successfully achieved lockup to an external time base, such as SMPTE time code or the optical input.

Address

The Address light glows continuously when the 896mk3 has successfully achieved lockup to SMPTE time code.

Freewheel

The Freewheel light illuminates when the 896mk3 is freewheeling address (time code), clock or both. For details about Freewheeling, see “Freewheel Address” and “Freewheel clock” below.

SMPTE source

Choose the analog input that is connected to the time code source. This is the input that the 896mk3 “listens” to for time code.

Freewheel Address

Freewheeling occurs when there is a glitch or drop-out in the incoming time code for some reason. The 896mk3 can freewheel past the drop-out and then resume lockup again as soon as it receives readable time code. Choose the amount of time you would like the 896mk3 to freewheel before it gives up and stops altogether.

The 896mk3 cannot freewheel address without clock. Therefore, the *Freewheel Address* setting will always be lower than or equal to the *Freewheel Clock* setting, and both menus will update as needed, depending on what you choose.

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

The ‘Infinite’ freewheel setting

The *Infinite* freewheel setting in the *Freewheel Address* menu causes the 896mk3 to freewheel indefinitely, until it receives readable time code again. To make it stop, click the *Stop Freewheeling* button.

Freewheel clock

Freewheeling occurs when there is glitch or drop-out in the incoming SMPTE time code for some reason. The 896mk3 can freewheel past the drop-out and then resume lockup again as soon as it receives a stable, readable clock signal.

The 896mk3 cannot freewheel address without clock. Therefore, the *Freewheel Address* setting will always be lower than or equal to the *Freewheel Clock* setting, and both menus will update as needed, depending on what you choose.

The ‘Infinite’ freewheel setting

The *Infinite* freewheel setting in the *Freewheel Clock* menu causes the 896mk3 to freewheel indefinitely, until it receives readable time code again. To make it stop, click the *Stop Freewheeling* button.

Stop Freewheeling

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

GENERATOR SECTION

The Generator section (on the right-hand side of the window in Figure 10-1) provides settings for generating SMPTE time code.

Level

Turn the level knob to adjust the volume of the SMPTE time code being generated by the 896mk3. The level knob disappears when the Destination is set to *None*.

Tach light

The Tach light blinks once per second when the 896mk3 is generating SMPTE time code.

Destination

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

Stripe

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

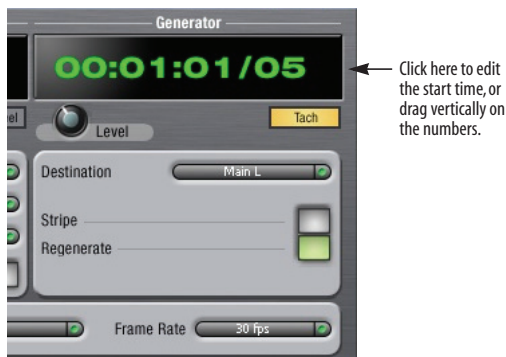


Figure 10-2: Setting the time code start time.

Regenerate

This option, when enabled, causes the generator to generate time code whenever the 896mk3 is receiving SMPTE time code.

SETTING UP FOR SMPTE TIME CODE SYNC

To set up direct SMPTE time code synchronization, see “Syncing to SMPTE time code” on page 34.

RESOLVING CUBASE OR NUENDO TO TIME CODE

To resolve your 896mk3 and Cubase or Nuendo directly to SMPTE time code with no additional synchronization devices, use the setup shown in “Syncing to SMPTE time code” on page 34. Make sure the *Master Clock Source* setting in the MOTU Audio Console window is set to *SMPTE*. Also, make sure that you’ve connected an LTC input signal to an 896mk3 analog input, and that you’ve specified that input in the *SMPTE Source* menu in SMPTE Console.

Part 3

Appendices

APPENDIX A **Audio I/O reference**

OVERVIEW

The MOTU Audio drivers supply text string labels for the 896mk3’s audio inputs and outputs to clearly identify each one, but some applications do not display these labels.

The following sections show how you can identify each input and output in a numbered list like this.

Windows: ASIO 125
Windows: WDM..... 126

WINDOWS: ASIO

Channels appear in the following order for ASIO applications on Windows.

Inputs at 1x sample rates

Inputs are always listed in the same order as follows, when operating the 896mk3 at 1x sample rates (44.1 or 48 kHz):

Input 44.1/48 kHz	Channels	List position	Comment
Analog	8	1-8	-
AES/EBU	2	9-10	-
SPDIF	2	11-12	-
Optical A	8 ADAT 2 TOSLINK	13-20 13-14	-
Optical B	8 ADAT 2 TOSLINK	21-28 15-16	These starting channel numbers assume that optical A is operating with the same format.
Stereo return	2	29-30 17-18	See “Return Assign” on page 58.
Reverb return	2	31-32 19-20	See “Reverb return” on page 58.

Inputs at 2x sample rates

When operating the 896mk3 at a 2x sample rate (88.2 or 96 kHz), inputs are listed as follows:

Input 44.1/48 kHz	Channels	List position	Comment
Analog	8	1-8	-
AES/EBU	2	9-10	-
SPDIF	2	11-12	-
Optical A	4 ADAT 2 TOSLINK	13-16 13-14	-
Optical B	4 ADAT 2 TOSLINK	17-20 15-16	These starting channel numbers assume that optical A is operating with the same format.
Stereo return	2	21-22 17-18	See “Return Assign” on page 58.
Reverb return	2	23-24 19-20	See “Reverb return” on page 58.

Inputs at 4x sample rates

When operating the 896mk3 at a 4x sample rate (176.4 or 192 kHz), inputs are listed as follows:

Input 176.4/192 kHz	Channels	List position	Comment
Analog	8	1-8	-
SPDIF	not available	not available	not available
AES/EBU	not available	not available	not available
Optical A	not available	not available	not available
Optical B	not available	not available	not available
Stereo return	2	9-10	See “Return Assign” on page 58.
Reverb return	not available	not available	not available

Outputs at 1x sample rates

Outputs are always listed in the same order as follows, when operating the 896mk3 at 1x sample rates (44.1 or 48 kHz):

Output 44.1/48 kHz	Channels	List position	Comment
Main outs	2	1-2	-
Analog	8	3-10	-
AES/EBU	2	11-12	-
SPDIF	2	13-14	-
Phones	2	15-16	-
Optical A	8 ADAT 2 TOSLINK	17-24 17-18	-
Optical B	8 ADAT 2 TOSLINK	25-32 19-20	These channel number ranges assume that optical A is operating with the same format.

Outputs at 2x sample rates

When operating the 896mk3 at a 2x sample rate (88.2 or 96 kHz), outputs are listed as follows:

Output 88.2/96 kHz	Channels	List position	Comment
Main outs	2	1-2	-
Analog	8	3-10	-
AES/EBU	2	11-12	-
SPDIF	2	13-14	-
Phones	Mirror only	--	-
Optical A	4 ADAT 2 TOSLINK	15-18 15-16	-
Optical B	4 ADAT 2 TOSLINK	19-22 17-18	These channel number ranges assume that optical A is operating with the same format.

Outputs at 4x sample rates

When operating the 896mk3 at a 4x sample rate (176.4 or 192 kHz), outputs are listed as follows:

Output 176.4/192 kHz	Channels	List position	Comment
Main outs	2	1-2	-
Analog	8	3-10	-
Phones	Mirror only	--	-
SPDIF	Not available	--	-
AES/EBU	Not available	--	-
Optical A	Not available	--	-
Optical B	Not available	--	-

WINDOWS: WDM

Channels appear in the following order for WDM applications on Windows.

Inputs at 1x sample rates

Inputs are always listed in the same order as follows, when operating the 896mk3 at 1x sample rates (44.1 or 48 kHz):

Input 44.1/48 kHz	Channels	List position	Comment
Stereo return	2	1-2	See "Return Assign" on page 58.
Analog	8	3-10	-
AES/EBU	2	11-12	-
SPDIF	2	13-14	-
Reverb return	2	15-16	See "Reverb return" on page 58.
Optical A	8 ADAT 2 TOSLINK	17-24 17-18	-
Optical B	8 ADAT 2 TOSLINK	25-32 19-20	These starting channel numbers assume that optical A is operating with the same format.

Inputs at 2x sample rates

When operating the 896mk3 at a 2x sample rate (88.2 or 96 kHz), inputs are listed as follows:

Input 88.2/96 kHz	Channels	List position	Comment
Stereo return	2	1-2	See “Return Assign” on page 58.
Analog	8	3-10	-
AES/EBU	2	11-12	-
SPDIF	2	13-14	-
Reverb return	not available	not available	not available
Optical A	4 ADAT 2 TOSLINK	15-18 15-16	-
Optical B	4 ADAT 2 TOSLINK	19-22 17-18	These starting channel numbers assume that optical A is operating with the same format.

Inputs at 4x sample rates

When operating the 896mk3 at a 4x sample rate (176.4 or 192 kHz), inputs are listed as follows:

Input 176.4/192 kHz	Channels	List position	Comment
Stereo return	2	1-2	See “Return Assign” on page 58.
Analog	8	3-10	-
AES/EBU	not available	not available	not available
SPDIF	not available	not available	not available
Reverb return	not available	not available	not available
Optical A	not available	not available	not available
Optical B	not available	not available	not available

Outputs at 1x sample rates

Outputs are always listed in the same order as follows, when operating the 896mk3 at 1x sample rates (44.1 or 48 kHz):

Output 44.1/48 kHz	Channels	List position	Comment
Main outs	2	1-2	-
Analog	8	3-10	-
AES/EBU	2	11-12	-
SPDIF	2	13-14	-
Phones	2	15-16	-
Optical A	8 ADAT 2 TOSLINK	17-24 17-18	-
Optical B	8 ADAT 2 TOSLINK	25-32 19-20	These channel number ranges assume that optical A is operating with the same format.

Outputs at 2x sample rates

When operating the 896mk3 at a 2x sample rate (88.2 or 96 kHz), outputs are listed as follows:

Output 88.2/96 kHz	Channels	List position	Comment
Main outs	2	1-2	-
Analog	8	3-10	-
AES/EBU	2	11-12	-
SPDIF	2	13-14	-
Phones	Mirror only	--	-
Optical A	4 ADAT 2 TOSLINK	15-18 15-16	-
Optical B	4 ADAT 2 TOSLINK	19-22 17-18	These channel number ranges assume that optical A is operating with the same format.

Outputs at 4x sample rates

When operating the 896mk3 at a 4x sample rate (176.4 or 192 kHz), outputs are listed as follows:

Output 176.4/ 192 kHz Channels		List position	Comment
Main outs	2	1-2	-
Analog	8	3-10	-
Phones	Mirror only	--	-
SPDIF	Not available	--	-
AES/EBU	Not available	--	-
Optical A	Not available	--	-
Optical B	Not available	--	-

APPENDIX B **Troubleshooting**

Why does the start sound not play through the 896mk3?

The 896mk3 will only playback audio at a sampling rate of 44.1, 48, 88.2 or 96 kHz.

The computer freezes when it starts up

If the computer is unable to boot up, it may be a conflict with the WDM Driver. As soon as the computer boots up, Windows will try and initialize the WDM Driver. If this fails, your computer will hang. To determine if the WDM Driver is the problem, boot up in Safe Mode or remove the MOTU Audio software with the Add/Remove Programs Control Panel then restart. If that fixes the problem, visit motu.com to download the very latest drivers and then install them.

'New hardware detected' window

If you connect your 896mk3 before running the MOTU Audio installer, Windows will prompt you that new hardware has been detected. Cancel this, and run the 896mk3 Software Installer, rather than allowing Windows to locate the drivers.

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 "Making sync connections" on page 28 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 due to hard drive problems

If you have checked your clock settings and you are still getting clicks and pops in your audio, you may have a drive related problem. Set your Master Clock Source to *Internal* and try recording just using the analog inputs and outputs of the 896mk3. If you encounter the same artifacts you may want

try using another drive in your computer. Clicks and pops can also occur when the drive is severely fragmented or there are other drive-related issues. If you are using a FireWire drive on the same bus as the 896mk3, it could be that the FireWire bus is overloaded (too many devices on the same bus). Try removing all devices except the 896mk3.

Connecting or powering gear during operation

It is not recommended that you connect/disconnect, or power on/off devices connected to the 896mk3 while recording or playing back audio. Doing so may cause a brief glitch in the audio.

896mk3 inputs and outputs are not available in host audio software

Make sure that the inputs and outputs are enabled. See "Working with 896mk3 inputs and outputs" on page 62.

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 Audio Console.

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 896mk3's hardware-based CueMix FX monitoring feature. Please see chapter 8, "Reducing Monitoring Latency" (page 65).

Controlling monitoring latency

See chapter 8, "Reducing Monitoring Latency" (page 65).

CUSTOMER SUPPORT

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

REPLACING DISCS

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

TECHNICAL SUPPORT

If you are unable, with your dealer's help, to solve problems you encounter with the 896mk3 system, you may contact our technical support department in one of the following ways:

- Tech support hotline: (617) 576-3066 (Monday through Friday, 9 a.m. to 6 p.m. EST)
- Online support: www.motu.com/support

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

- The serial number of the 896mk3 system. This is printed on a label placed on the bottom of the 896mk3 rack unit. You must be able to supply this number to receive technical support.
- A brief explanation of the problem, including the exact sequence of actions which cause it, and the contents of any error messages which appear on the screen.
- The pages in the manual which refer to the parts of the 896mk3 or AudioDesk with which you are having trouble.
- The version of your computer's operating system.

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

If you have features or ideas you would like to see implemented, we'd like to hear from you. Please write to the 896mk3 Development Team, MOTU Inc., 1280 Massachusetts Avenue, Cambridge, MA 02138 or send an email to: customerservice@motu.com.

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