828x™

User Guide for Mac

THUNDERBOLT™

MOTU

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

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

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

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

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

CE

Contents

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 828x COULD CAUSE AN ELECTRICAL SHOCK.

The MOTU 828x 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 828x 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

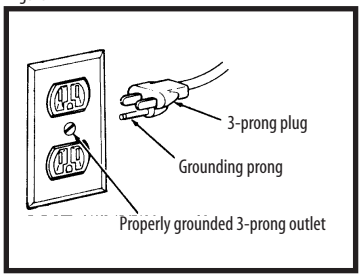
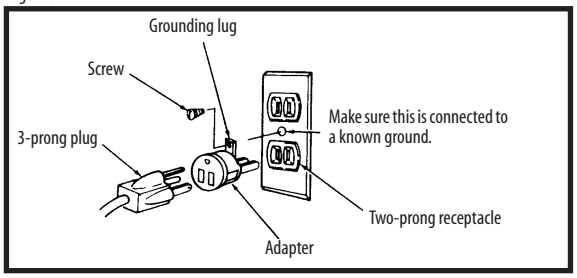


Figure B



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 828x plug.

IMPORTANT SAFEGUARDS

1. Read these instructions. All the safety and operating instructions should be read before operating the 828x.
2. Keep these instructions. These safety instructions and the 828x owner's manual should be retained for future reference.
3. Heed all warnings. All warnings on the 828x 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 828x near water.
6. Cleaning - Unplug the 828x 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 828x 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 and electrician for replacement of the obsolete outlet.
11. Power cord - Protect the 828x 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 828x 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 828x 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 828x. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
16. Surge protection - Unplug the 828x 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 828x 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 828x, the 828x 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 828x 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 828x, 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°)

TO REDUCE THE RISK OF ELECTRICAL SHOCK OR FIRE

CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN

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 • 20 Watts.

Part 1
Getting Started

Quick Reference: 828x Front Panel

1. These XLR/TRS combo jacks accept either a mic cable or a quarter-inch guitar cable. Both the low-impedance XLR jack and the high-impedance TRS jack are equipped with a preamp (so don't connect a +4 line level XLR cable!)

2. The upper phone jack 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 MAIN VOL knob next to it controls the main outs and this jack, but MAIN VOL can be programmed to control any combination of outputs. See "The Monitor Group" on page 91 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 outs, the MAIN VOL knob will no longer control the volume of this phone jack, either.

3. These two trim knobs provide approximately 53 dB of gain for the XLR mic input and the hi-Z TRS guitar/instrument input. Both inputs have preamps, so you can plug just about anything into them: a microphone, a guitar, a synth — but don't plug in a +4 signal here (due to the preamps): use a rear-panel TRS input instead. Use the trim knob and the "MIC" input level meters over in the metering section to calibrate the input signal level. The meters cover both the TRS and XLR input. These mic inputs are also equipped with the 828x'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 46 for details. Use the rear panel sends to route these inputs to your favorite outboard gear. Use any rear-panel input as a return.

4. This section controls the 828x'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, compression, and reverb to inputs, outputs and mix busses. The four knobs to the left of 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 mix settings for the chosen channel.

5. This bank of four-segment input meters is for the 8 analog TRS input jacks on the rear panel.

6. Four-segment stereo metering for S/PDIF input.

7. This section provides two ten-segment meters for the two front-panel mic/guitar inputs. The meters show input levels from -42 to -1 in the first column of LEDs, plus an additional range in a second column from zero to +12 dB (including clip). Both 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 turned on, 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.

8. Five-segment metering for the main outs. Use the MAIN VOL knob to control output level.

9. These lights indicate the global sample rate at which the 828x is operating. Use the MOTU Audio Setup software to set the sample rate or to choose an external clock source, from which the sample rate will be derived. When no clock signal is currently present, one of these LEDs flashes rapidly.

10. Thunderbolt and USB are "plug-and-play" protocols. That means that you can turn off the 828x and turn it back on without restarting your computer.

11. When the 828x is resolving to SMPTE time code, the LOCK/TACH LED glows when lockup has been achieved. The ADAT and MIDI LEDs blink when there is optical audio or MIDI activity, respectively.

12. These round LEDs indicate signal presence on the 8 rear-panel TRS analog and SPDIF outputs. Their threshold is around -42 dB. They do not indicate clipping in any way; use your host audio software level meters to calibrate output levels.

13. The multi-purpose backlit LCD shows system settings or CueMix FX settings, depending on which knobs you turn. The labels above and below the LCD refer to the four digital rotary encoders to the left of the LCD. These knobs let you access and program all settings in the 828x.

14. 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 828x settings, such as the global sample rate, etc.

15. Use these buttons to toggle the -20 dB pad and 48V phantom power for each mic input. The Precision Digital Trim™ knob provides 53 dB of gain.

16. From the factory, the PHONES jack is a discrete output (when the 828x is operating 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 outs. As the primary phone jack, it has its own dedicated volume knob (to its right).

Quick Reference: 828x Rear Panel

1. The 828x is equipped with an auto-switching international power supply.

2. These are standard BNC word clock jacks. Use them for a variety of applications, such as for digital transfers with devices that cannot slave to the clock supplied by their digital I/O connection with the 828x.

3. These jacks provide stereo, 24-bit S/PDIF digital input and output at all supported sample rates (up to 96 kHz).

4. Connect a MIDI device here using standard MIDI cables. Connect the 828x's MIDI OUT port to the MIDI IN port on the other device. Conversely, connect the 828x's MIDI IN port to the MIDI OUT port on the other device. You can connect different devices to each port, such as a controller device to the IN port and a sound module to the OUT port. You can also daisy-chain MIDI devices, but be sure to manage their MIDI channels (so that they don't receive or transmit on the same channel).

5. 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 Setup software (or using the front panel LCD). (See "Optical input/output" on page 43) 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.

   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).

6. Equipped with 24-bit 192 kHz converters, these 8 analog inputs are balanced TRS (tip/ring/sleeve) quarter-inch connectors that can also accept an unbalanced plug. They do not have microphone preamps, so they are best used for synthesizers, drum machines, effects processors, and other instruments with line level signals (either -10 dB or +4 dB). These inputs are also equipped with the 828x's Precision Digital Trim™ feature: digitally controlled analog trims that let you adjust input level in approximately 1 dB increments from either front panel LCD or the included CueMix FX software. The trim can be adjusted over a range of -96 to +22 dB.

7. The 828x's eight analog outputs are balanced +4dB TRS (tip/ring/sleeve) quarter-inch connectors that can also accept an unbalanced plug. They are equipped with 24-bit 192 kHz converters.

8. These are quarter-inch analog SMPTE input and output jacks. Use them to resolve the 828x directly to time code and transmit time code to other devices.

9. These two XLR jacks serve as the 828x's main outputs. You can connect them to a set of powered studio monitors and then control the volume from the front panel MAIN VOL knob.

   To hear audio playback from your host audio software on these main outs, assign the audio tracks (and master fader) to these main outs. You can also use CueMix FX to monitor live 828x inputs here as well.

10. These two quarter-inch balanced TRS send outputs supply the pre amplified input signal from the mic/guitar/instrument inputs on the front panel. Use them to insert your favorite compressor, EQ, reverb or other outboard effect. Use any TRS input as a return.

11. 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 44.

12. Connect the 828x to the computer here via either Thunderbolt or USB 2.0, using a standard Thunderbolt or USB cable. If you use Thunderbolt, you can daisy-chain up to six Thunderbolt devices together, with the 828x at the end of the chain (because it has one Thunderbolt port). For details, see "Connecting multiple 828x interfaces" on page 35.

Quick Reference: MOTU Audio Setup

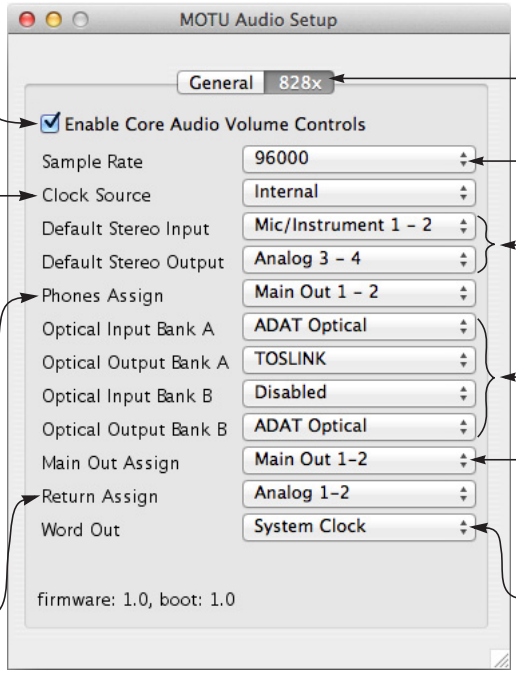
Lets you to use the Mac's volume controls to control the output level of the 828x when it is being used as the audio output hardware for your Mac.

Determines the clock source for your 828x. If you're just using the analog ins and outs, set this to 'Internal'. The other settings are for digital transfers via S/PDIF or optical ports, or for slaving the 828x to word clock.

- ✓ Internal
- Word Clock In
- S/PDIF
- ADAT Optical A
- SMPTE
- Built-in Audio

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.)

The 828x driver provides a stereo return back to the computer. This return feeds the signal on any 828x 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 828x only, back into the computer.



Click the tabs to access general MOTU interface settings or settings specific to the 828x (or other connected interface).

Choose the global sample rate for the system here.

Specifies the stereo input and output pair when the 828x is chosen for Mac OS X audio I/O.

Each optical bank can be configured independently for ADAT or TOSLink. Disable them when not in use to conserve DSP and bus bandwidth.

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

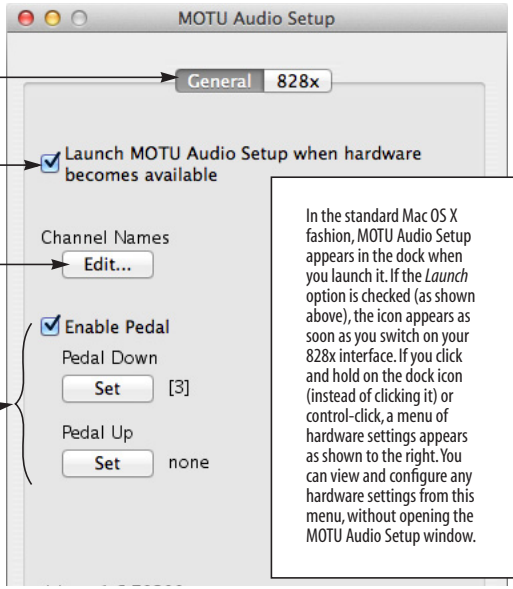
If you are running an 828x interface at a high sample rate (88.2, 96, 176.4 or 192 kHz), this option appears in the interface tab. It lets you choose a word clock output rate that either matches the global sample rate (e.g. 96kHz) or reduces it to the corresponding 1x rate (e.g. 48kHz instead of 192 kHz).

Click the General tab to access these settings.

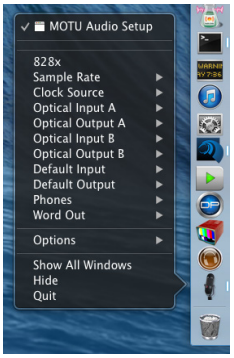
Check this option if you would like the MOTU Audio Setup icon to appear in the application dock as soon as a MOTU interface is detected (switched on, plugged in, etc.)

This button opens another dialog that lets you assign your own customized names to each 828x input and output. For example, if you have a lead vocal mic plugged into input 1, you could name it "Lead Vox". Your customized names then appear in your host audio application (if it supports Core Audio input naming).

If you have a foot switch connected to the 828x, 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 44.



In the standard Mac OS X fashion, MOTU Audio Setup appears in the dock when you launch it. If the *Launch* option is checked (as shown above), the icon appears as soon as you switch on your 828x interface. If you click and hold on the dock icon (instead of clicking it) or control-click, a menu of hardware settings appears as shown to the right. You can view and configure any hardware settings from this menu, without opening the MOTU Audio Setup window.



CHAPTER 1 About the 828x

OVERVIEW

The 828x is a hybrid Thunderbolt™/USB2 audio interface for Mac and Windows with on-board effects and mixing that offers 28 inputs and 30 outputs at 44.1 or 48 kHz. 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 828x consists of a standard 19-inch, single-space, rack-mountable I/O unit that connects directly to a computer via a standard Thunderbolt or USB cable.

The 828x offers the following main features:

- Universal computer connectivity with Thunderbolt or high-speed USB2
- Eight 24-bit analog quarter-inch (TRS) inputs
- Eight 24-bit analog quarter-inch (TRS) outputs
- Two combo XLR/TRS mic/guitar inputs with preamps, individual sends, 48V phantom power, 20 dB pad, and Precision Digital Trim™
- Two XLR main outputs
- Operation on all analog I/O at standard sample rates up to 192 kHz
- Digitally controlled analog trim for all analog inputs
- 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
- RCA S/PDIF at sample rates up to 96 kHz
- Word clock I/O
- MIDI I/O
- On-board SMPTE synchronization with dedicated SMPTE I/O jacks
- Foot switch for hands-free punch-in/out
- Two headphone jacks with independent volume control
- Programmable main volume knob
- CueMix™ FX no-latency mixing, monitoring and effects processing
- Front-panel LCD programming for the mixer and all other settings
- Extensive front panel metering and status LEDs
- 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
- AudioDesk™, full-featured audio workstation software

With a variety of I/O formats, mic preamps, no-latency mixing and processing of live input and synchronization capabilities, the 828x is a complete, portable "studio in a box" when used with a Mac or Windows computer.

THE 828X REAR PANEL

The 828x rear panel has the following connectors:

- Eight balanced quarter-inch (TRS) analog outputs (with 24-bit 192 kHz converters)
- Eight balanced quarter-inch (TRS) analog inputs (with 24-bit 192 kHz converters)
- Two XLR "main" analog outputs with 24-bit 192 kHz converters
- Two balanced quarter-inch (TRS) analog sends (for the front-panel mic/guitar inputs)
- Balanced TRS quarter-inch analog in/out dedicated for SMPTE time code
- Two sets of optical connectors (in and out), individually switchable among ADAT optical "lightpipe", 96 kHz S/MUX optical or S/PDIF "TOSLink"
- RCA S/PDIF in/out
- MIDI IN and MIDI OUT
- Word clock in/out
- Foot pedal jack
- One Thunderbolt connector
- One high-speed USB2 connector

28 inputs and 30 outputs

All 828x inputs and outputs can be used simultaneously, for a total of 28 inputs and 30 outputs when operating at 44.1 or 48 kHz:

Connection	Input	Output
Analog 24-bit 192 kHz on bal/unbal TRS	8	8
Mic/guitar 24-bit 192 kHz on XLR/TRS combo	2	-
Main outputs 24-bit 192 kHz on XLR	-	stereo
Headphone output*	-	stereo
ADAT optical digital†	16	16
RCA S/PDIF 24-bit 96kHz digital	stereo	stereo
Total	28	30

* The phone jack next to the MAIN 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 828x output pair, such as the main outs.

† The 828x optical connectors support several standard optical I/O formats, which provide varying channel counts. See "Optical" on page 13 for details about optical bank operation.

With the exception of the phone jack on the front panel labeled "(MAIN)", all inputs and outputs are discrete. For example, using a mic input does not "steal" an input from the TRS analog I/O bank.

Analog

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

All quarter-inch analog inputs can accept either a balanced or unbalanced plug.

The quarter-inch outputs are referenced to a +4 dBu line level output signal. The inputs have +22 dB of input gain and -96 dB of cut, allowing them to accommodate both -10 dBu and +4 dBu level signals.

Precision Digital Trim™

All of the 828x's analog inputs are equipped with digitally controlled analog trims, adjustable in 1 dB increments. The mic/guitar 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, including eight rear-panel TRS analog inputs, can be trimmed using the front panel LCD or using the 828x's included CueMix FX control software for Mac and Windows. This gives you finely-tuned control of trim settings for synths, effects modules, and a wide variety of analog inputs for optimum levels. Different trim configurations can then be saved as preset configurations for instant recall.

Mic/guitar sends

Before A/D conversion, the pre-amplified signal from each front-panel mic/guitar input is routed to one of the two rear-panel quarter-inch analog sends, so that you can insert a favorite outboard EQ, compressor, amp or effects processor to the mic/guitar input signal before it is converted to digital form. The resulting output from the outboard gear can be fed back into the 828x via one of the eight TRS analog inputs on the rear panel, for routing to the computer and/or inclusion in the 828x's built-in monitor mixes.

Main Outs

The main outs are equipped with 24-bit 192 kHz D/A converters and serve as independent outputs for the computer or for the 828x's on-board CueMix FX mixes.

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.

S/PDIF

The 828x 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.

MIDI I/O

The 828x's standard MIDI IN and MIDI OUT jacks supply 16 channels of MIDI I/O to and from the computer via the 828x's Thunderbolt connection. Timing accuracy can be sample-accurate with host software that supports it.

On-board SMPTE synchronization

The 828x can resolve directly to SMPTE time code via the quarter-inch SMPTE input, without a separate synchronizer. A SMPTE out jack is also provided for time code generation. The 828x provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy.

The included MOTU SMPTE Setup™ software includes a complete set of tools for generating and regenerating SMPTE time code, providing a way to slave other devices to the computer.

Word clock

The 828x supports standard word clock synchronization at any supported sample rate. When the 828x is operating at 96 kHz, it can generate word clock output at either 96 or 48 kHz. Half-rate output is supported for all high sample rates (from 88.2 to 192 kHz).

Punch in/out

The quarter-inch Punch in/out jack accepts a standard foot switch. When you push the foot switch, the 828x triggers a programmable keystroke on the computer keyboard. For example, with MOTU's Digital Performer audio sequencer software, the foot switch triggers the 3 key on the numeric keypad, which toggles recording in Digital Performer. Therefore, pressing the foot switch is the same as pressing the 3 key. The 828x Control Panel software lets you program any keystroke you wish.

Hybrid Thunderbolt/USB 2.0 connectivity

Thunderbolt is a new, high-performance, high-bandwidth connectivity standard for Mac and Windows computers. High-speed USB 2.0 is a widely adopted standard for connecting peripheral devices to personal computers.

To fully support both formats, your 828x audio interface is equipped with both a Thunderbolt connector and a high-speed USB 2.0 connector, and you can use either port (one or the other) to connect the 828x to your computer. This gives you maximum flexibility and compatibility with today's ever-expanding universe of Mac and Windows computers.

THE 828X FRONT PANEL

Mic/guitar inputs with preamps

The two mic/instrument inputs (front panel and rear panel) are equipped with preamps and "combo" XLR/TRS jacks, which accept XLR microphone inputs or quarter-inch guitar/instruments inputs. Individual 48 volt phantom power and a 20 dB pad can be supplied independently to each mic input. The Precision Digital Trim™ knobs on the front panel for each mic/instrument input provide up to 53 dB of boost in precise 1 dB increments.

As explained in "Mic/guitar sends" on page 13, the pre-amplified signal can be routed to external outboard gear before being routed back into the 828x.

Mic/guitar input overload protection

Both mic/guitar 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 828x's Soft Clip feature, which engages just before clipping occurs and helps reduce perceptible distortion.

Headphone output and main volume control

The 828x 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 MAIN VOL knob can be programmed to control any combination of outputs (analog or digital). For example, it can control monitor output for an entire 5.1 or 7.1 surround mix.

Programmable backlit LCD display

Any 828x 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.

Metering section

The front panel of the 828x displays several banks of input and output metering. The threshold for these lights is approximately -42 dB. The four- and five-segment input meters provide dedicated multi-segment metering for their respective inputs, as do the five-segment main out meters.

Two ten-segment meters for the two front-panel mic/guitar inputs show input levels from -42 to -1 in the first column of LEDs, plus an additional range in a second column from zero to +12 dB (including clip). Both 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 turned on, 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.

The *Clock* lights indicate the global sample rate (as chosen in the MOTU Audio Setup software). The LOCK and TACH LEDs provide feedback for the 828x's on-board SMPTE synchronization features. The ADAT and MIDI LEDs indicate audio and MIDI activity, respectively.

16-BIT AND 24-BIT RECORDING

The 828x 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 828x'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 828x 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 828x hardware, independent of the computer. Effects can even be applied when the 828x 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 828x'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 828x'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 (included) allows you to keep tabs on the 828x'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.

AUDIODESK

AudioDesk is a full-featured, 24-bit audio workstation software package included with the 828x system (for Mac OS X only). AudioDesk provides multi-channel waveform editing, automated virtual mixing, graphic editing of ramp automation, real-time effects plug-ins with 32-bit floating point processing, crossfades, support for many third-party audio plug-ins, background processing of file-based operations, sample-accurate editing and placement of audio, and more.

DIGITAL PERFORMER

The 828x system is fully integrated with MOTU's award-winning Digital Performer audio sequencer software package.

OTHER HOST AUDIO SOFTWARE

The 828x system includes a standard Mac OS X CoreAudio driver for multichannel I/O with any audio application that supports CoreAudio.

CHAPTER 2 Packing List and System Requirements

PACKING LIST

The 828x ships with the items listed below. If any of these items are not present in your 828x box when you first open it, please immediately contact your dealer or MOTU.

- One 828x
- One USB cable
- One power cord
- One 828x Mac/Windows manual
- One cross-platform installer disc
- Product registration card

MAC SYSTEM REQUIREMENTS

The 828x system requires the following Mac system:

- PowerPC G4 CPU 1 GHz or faster (including PowerPC G5 CPUs and all Intel processor Macs)
- 1 GB RAM; 2 GB or more recommended
- Mac OS X version 10.5.8 or later required
- Available Thunderbolt or high-speed USB 2.0 port
- A large hard drive (preferably at least 250 GB)

PLEASE REGISTER TODAY!

Please register your 828x 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.

Be sure to do the same for the included AudioDesk software, which must be registered separately. You can do so online or by filling out and mailing the included software registration card found at the beginning of your AudioDesk manual. Please be sure to register AudioDesk as well, so that you will be eligible to receive technical support and announcements about AudioDesk software enhancements as soon as they become available.

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

CHAPTER 3 Installing the 828x Software

OVERVIEW

INSTALLATION

Before you connect the 828x to your computer and power it on, run the 828x software installer. This ensures that all the 828x components are properly installed in your system.

Run the MOTU Audio installer

Install the 828x 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 *MOTU Audio Installer* application.

4 Follow the directions that the installer gives you.

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

MOTU AUDIO DRIVER

Core Audio is a term that refers to the software technology built into Mac OS X that provides all of its standardized audio features. More specifically, we use *Core Audio* to refer to Mac OS X's standard audio driver model. The *MOTU Core Audio driver* provides multi-channel audio input and output with any Mac OS X Core Audio-compatible software.

Software component	Purpose	For more information
MOTU Audio and MIDI drivers	Provides multi-channel audio input and output for MOTU Thunderbolt, FireWire and USB Audio devices with host audio software.	"MOTU Audio driver" on page 19
MOTU Audio Setup	Provides access to all of the settings in the 828x and other MOTU interfaces. Required for 828x operation.	chapter 5, "MOTU Audio Setup" (page 39)
CueMix FX	Gives you complete control over the 828x's CueMix FX on-board mixer, which provides no-latency monitoring, mixing and processing of live inputs through your 828x.	chapter 9, "CueMix FX" (page 69)
MOTU SMPTE Setup	Provides access to the 828x system's SMPTE time code sync features.	chapter 10, "MOTU SMPTE Setup" (page 115)
AudioDesk	Provides complete multi-track recording, mixing and processing. Optional.	AudioDesk User Guide

For details about using the 828x with Core Audio applications, see chapter 7, "Configuring Host Audio Software" (page 55).

MOTU AUDIO SETUP

MOTU Audio Setup (available in the Applications folder) gives you access to all of the settings in the 828x, such as the clock source and sample rate. For complete details, see chapter 5, "MOTU Audio Setup" (page 39).

CORE MIDI AND AUDIO MIDI SETUP

Core MIDI is the "under-the-hood" portion of Mac OS X that handles MIDI services for MIDI hardware and software. Core MIDI provides many universal MIDI system management features, including MIDI communication between your 828x interface and all Core MIDI compatible software.

Audio MIDI Setup is a utility included with Mac OS X that allows you to configure your 828x interface for use with all Core MIDI compatible applications. Audio MIDI Setup provides:

- A "virtual" studio on your Mac that graphically represents your MIDI hardware setup and that is shared by all Core MIDI-compatible programs
- A simple, intuitive list of your MIDI devices whenever you need it in any Core MIDI-compatible program

Launching Audio MIDI Setup

1 Make sure your 828x interface is connected and turned on.

2 Launch the Audio MIDI Setup utility.

This can usually be found in /Applications/Utilities. If it has been moved, just search for *Audio MIDI Setup*.

3 Confirm that the MIDI interface is present in the *MIDI Devices* tab (or window) in Audio MIDI Setup.

If the interface does not appear, or if it is grayed out, check your cable connections and click *Rescan MIDI*.

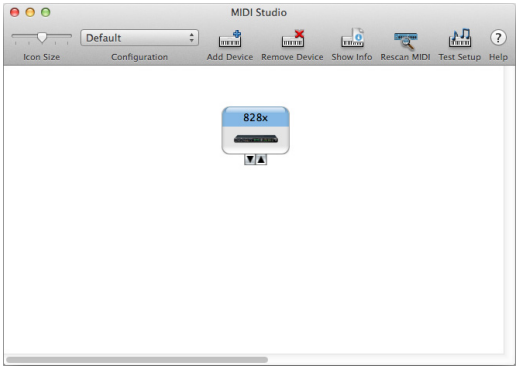


Figure 3-1: The 828x interface as it appears in the MIDI tab of Audio MIDI Setup.

Connecting MIDI devices to the 828x

Once your 828x interface appears in Audio MIDI Setup, you are ready to add devices, indicate how they are connected, and identify properties they may have for particular purposes. This information is shared with all Core MIDI compatible applications.

To add a device in Audio MIDI Setup:

1 Click *Add Device*.

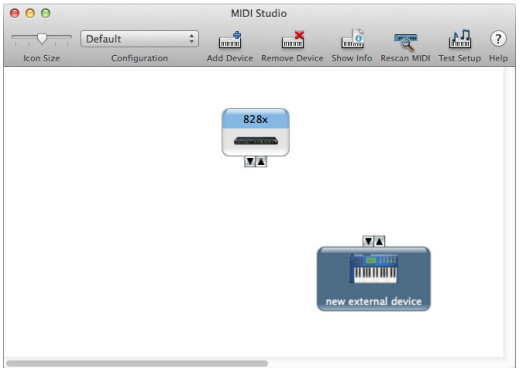


Figure 3-2: Adding a MIDI device.

2 Drag on its input and output arrows to draw connections to the 828x that match its physical connection.

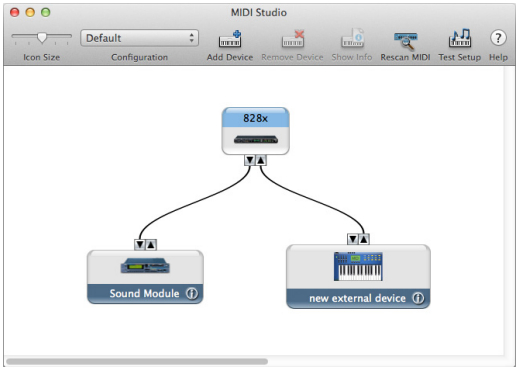


Figure 3-3: Connecting devices to the 828x. In this example, a controller keyboard is connected to the 828x's MIDI IN, and a sound module is connected to the 828x MIDI OUT.

3 Double-click the device to make settings, such as input and output channels, that further describe the device.

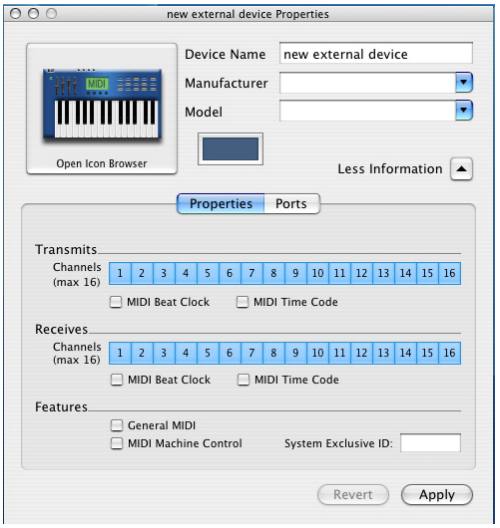


Figure 3-4: Device settings.

4 Repeat the above steps for each MIDI device connected to the interface.

5 When you are finished, quit Audio MIDI Setup.

Your configuration is automatically saved as the default configuration, and it is shared with all Core MIDI-compatible software.

CUEMIX FX

CueMix FX (available in the Applications folder) provides control over the 828x's no-latency CueMix FX on-board mixing, effects processing, an instrument tuner, a full-featured oscilloscope, and other audio analysis tools. For details, see chapter 9, "CueMix FX" (page 69).

MOTU SMPTE SETUP

MOTU SMPTE Setup (available in the Applications folder) software provides a complete set of tools to resolve the 828x 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 Setup" (page 115).

AUDIODESK WORKSTATION SOFTWARE

AudioDesk is an advanced workstation software package for the 828x that lets you record, edit, mix, process, bounce and master multi-track digital audio recording projects. Advanced features include real-time 32-bit effects processing, 24-bit recording, and much more.

See the *AudioDesk User Guide* included with your 828x system for details.

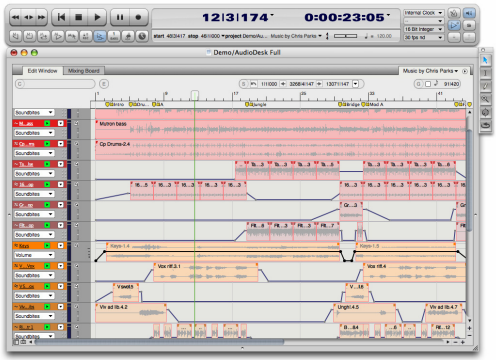


Figure 3-5: AudioDesk for Mac OS X.

CHAPTER 4 Installing the 828x Hardware

OVERVIEW

Here's an overview for installing the 828x:

CONNECT THE 828X INTERFACE

Your 828x audio interface is equipped with both a Thunderbolt connector and a high-speed USB 2.0 connector, and you can use either port (one or the other) to connect the 828x to your computer. This gives you maximum flexibility and compatibility with today's ever-expanding universe of Mac and Windows computers.

The 828x is a Thunderbolt 1 device. It is compatible with Thunderbolt 1 and Thunderbolt 2.

Should I use Thunderbolt or USB 2.0?

If your computer has both Thunderbolt and USB 2.0, then it is your choice, and your decision may depend mostly on other peripherals you may have.

If your computer does not have a Thunderbolt port, then obviously you will need to connect the 828x to one of its high-speed USB 2.0 ports.

If you are connecting with Thunderbolt

Using a standard Thunderbolt cable, connect one end of the cable to the Thunderbolt socket on the 828x, and connect the other end to any available Thunderbolt port on the computer.

Connecting multiple Thunderbolt devices

Thunderbolt allows you to connect multiple devices to a host computer, through multiple Thunderbolt ports on the host, or by daisy-chaining up to six devices from a single host Thunderbolt port. Since the 828x has one Thunderbolt port on it, place it at the end of the daisy chain.

Thunderbolt is designed to provide enough bandwidth to easily support the 828x and other Thunderbolt devices, such as hard drives and

displays. You should not hesitate to connect other Thunderbolt devices like these to your computer, along with the 828x, as your needs require.

Also see "Connecting multiple 828x interfaces" on page 35.

USB 3.0, USB 2.0 and USB 1.1

There are primarily three 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 3.0 and 2.0 controllers support high speed devices such as the 828x. Since the 828x requires a high speed connection, it must be connected to a USB 3.0 or 2.0 host controller or hub.

For the most reliable connection, it is recommended that you connect the 828x directly to one of your computer's USB 2.0-or 3.0-compatible ports. However, since USB 3.0 and 2.0 hubs are compatible with both types of devices, the 828x can be connected to a USB 3.0/2.0 hub along with USB 1.1 devices if necessary. The 828x 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, 2.0, or 3.0:

1 From the Apple menu, choose About this Mac.

2 Click More Info.

3 Click System Report.

4 In the Hardware section of the System Report, click USB.

5 You will see a section for each USB bus in the USB Device Tree window. USB 3.0 ports will appear in the *USB 3.0 SuperSpeed* bus, USB 2.0 ports appear as *USB Hi-Speed* bus, and USB 1.0 ports will be identified by *USB Bus*. Click the disclosure triangle to see more detail on the devices connected.

If you are connecting with USB 2.0 or 3.0

1 Before you begin, make sure your computer and the 828x are switched off.

2 Plug the flat "type A" plug of the 828x USB cable (included) into a USB2-equipped socket on the computer.

3 Plug the squared "type B" plug of the USB cable into the 828x I/O.

CONNECT AUDIO INPUTS AND OUTPUTS

The 828x audio interface has the following audio input and output connectors:

- 8 balanced, +4 dB quarter-inch analog outputs
- 8 balanced +4 dB quarter-inch analog inputs
- 2 mic/guitar combo jack inputs with preamps
- 2 quarter-inch sends for the mic/guitar inputs
- 2 XLR main outs
- 2 pairs of optical in/out banks switchable between ADAT ("Lightpipe") or optical S/PDIF (TOSLink)
- 1 pair of RCA S/PDIF in and out

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

Mic/guitar inputs with preamps

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

Figure 4-1: Mic/guitar inputs.

☛ Do not connect a +4 (line level) XLR cable to the front-panel inputs (because of the preamps). Use a rear-panel quarter-inch input instead.

Phantom power

If you are connecting a condenser microphone or another device that requires phantom power, engage the corresponding front-panel phantom power switch.

Trim

Both the low-impedance XLR mic input and the high-impedance quarter-inch guitar input are equipped with 53 dB of digitally controlled analog trim. Use the detented trim knobs to adjust the input level as needed for each input. The LCD provides visual feedback as you turn the trim knob.

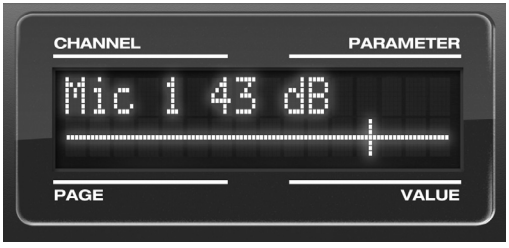


Figure 4-3: The LCD gives you feedback as you turn the TRIM knobs for the two mic/guitar inputs.

The 828x's input trims are digitally controlled, so they allow you to make fine-tuned adjustments in approximately 1dB increments. You can also adjust trim in the MOTU CueMix FX software. See "Input trim" on page 75.

-20 dB pad

Each mic input (XLR jack) is equipped with a -20 dB pad switch, to accommodate input signals that could overdrive the input.

☛ The pad is not available for the TRS jack.

Figure 4-2: 828x front panel

Combo jack summary

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

Input	48V	Pad	Trim
Condenser mic	On	As needed	As needed
Dynamic mic	Off	As needed	As needed
Guitar	Off	n/a	As needed
-10 dB Line level via TRS	Off	n/a	As needed
-10 dB Line level via XLR	Off	-20 dB	+12dB
+4 dB line level (XLR only)	Off	-20 dB	Zero

Quarter-inch analog

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

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

Quarter-inch analog input trims

The quarter-inch inputs are calibrated to accommodate either +4 or -10 dBu signals and are equipped with digitally controlled analog trims that provide +22 dB of gain and -96 dB of cut. You can use either the front panel LCD or the included CueMix FX software to adjust the input trim. To adjust these trims using CueMix FX, see "Input trim" on page 75. To adjust the trims using the front panel LCD:

1 Push the CHANNEL knob repeatedly until you see "I:" (which stands for *Input*) in the CHANNEL section of the LCD (Figure 4-5).

2 Turn the CHANNEL KNOB until you see the desired analog input or output pair. For example, analog inputs 1-2 appear as "I:An 1-2" (Figure 4-5), which means *Input analog 1-2*.

3 From the factory, analog inputs are grouped in stereo pairs (1-2, 3-4, etc.) If you need to split a pair to deal with it as two individual mono inputs, turn the PARAMETER knob until you see PAIR in the parameter section of the LCD (Figure 4-5). Turn the VALUE knob to choose MONO. Then turn the CHANNEL knob again to select the desired input you are adjusting.

Figure 4-5: The settings for analog inputs 1 and 2 (as a pair).

4 After splitting the stereo pair, if necessary, turn the PARAM knob until you see the TRIM parameter in the LCD (Figure 4-6):

Figure 4-4: 828x back panel

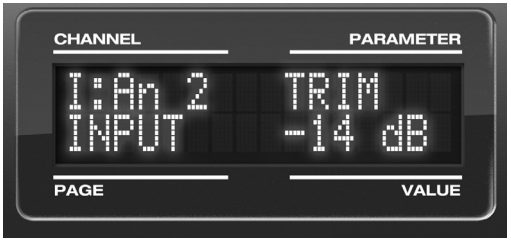


Figure 4-6: Setting the input trim for a TRS analog input pair.

5 Turn the VALUE knob to adjust the trim.

Mic/guitar/instrument sends

Each front-panel XLR/TRS input has a corresponding send on the rear panel (Figure 4-4). The output from this send is the pre-amplified and calibrated signal from the corresponding mic or guitar input, which you can then route to any other device, such as compressor, guitar amp, outboard EQ, reverb unit, etc. Use any 828x input (analog or digital) as a return back into the 828x. From there, you will be able to route the signal anywhere in the system, such as to the computer and/or to any CueMix FX mix bus.

XLR main outs

The XLR main outputs serve as independent outputs. From the factory, the main out volume is controlled by the MAIN 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 91. 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 828x rear panel provides two sets of ADAT optical ("lightpipe") connectors: Bank A and B (Figure 4-4). 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 828x supplies +12dB of digital trim (boost) for each optical input, which can be adjusted from CueMix FX ("Input trim" on page 75) or the front panel ("The IN (inputs) menu" on page 49).

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 48.

Using optical I/O to operate the 828x as a 16-channel expander

When the 828x is not connected directly to a computer via Thunderbolt or USB, the sixteen optical output channels can be programmed (via the CueMix FX mixer) to mirror the incoming signal on any combination of the 828x's inputs. By connecting the 828x optical outputs to another device, such as another ADAT-optical equipped interface or a digital mixer, you add up to sixteen additional inputs to your system (or eight inputs at the 2x sample rates).

To learn how to program the 828x when it is operating as a stand-alone expander in this fashion, see chapter 6, "Front Panel Operation" (page 45).

Choosing a clock source for optical connections

When connecting an *optical* device, make sure that its digital audio clock is phase-locked (in sync with) the 828x, as explained in "Making sync connections" on page 32. There are two ways to do this:

1. Resolve the optical device to the 828x
2. Resolve the 828x to the optical device

For 1), choose *Internal* (or any other clock source except *ADAT optical*) as the clock source for the 828x in MOTU Audio Setup.

For 2), choose either *ADAT Optical A* or *ADAT Optical B* as the 828x's clock source (Figure 4-7). Be sure to choose the optical port that the device is connected to.

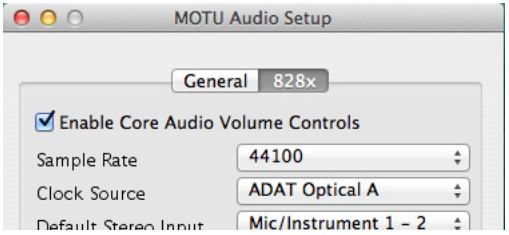


Figure 4-7: Resolving the 828x to an optical device.

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

Using word clock to resolve optical devices

If the optical device you are connecting to the 828x has word clock connectors on it, you can use them to resolve the device to the 828x, similar to the diagram shown in Figure 4-16 on page 34 for S/PDIF devices with word clock. Also see "Syncing word clock devices" on page 35.

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 34.

The 828x supplies +12dB of digital trim (boost) for the S/PDIF input pair, which can be adjusted from CueMix FX ("Input trim" on page 75) or the front panel ("The IN (inputs) menu" on page 49).

CONNECT MIDI GEAR

Connect your MIDI device's MIDI IN jack to the 828x's MIDI OUT jack (Connection A below). Conversely, connect the MIDI device's MIDI OUT jack to the 828x's MIDI IN jack (Connection B).

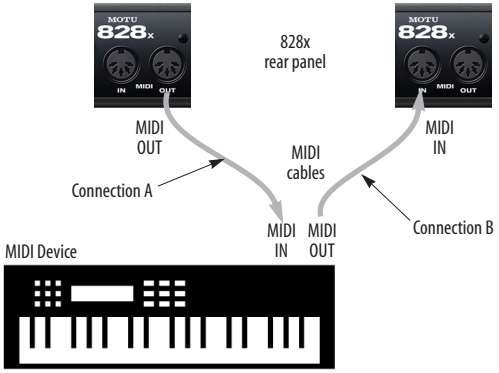


Figure 4-8: Connecting a MIDI device to the 828x.

One-way MIDI connections

MIDI devices that do not receive MIDI data, such as a dedicated keyboard controller, guitar controller, or drum pad, only need Connection B shown in Figure 4-8. Similarly, devices that never send data, such as a sound module, only need Connection A. Make both connections for any device that needs to both send and receive MIDI data.

Connecting additional gear with MIDI THRUs

If you need to connect several pieces of MIDI gear, run a MIDI cable from the MIDI THRU of a device already connected to the 828x to the MIDI IN on the additional device as shown below in Figure 4-9. The two devices then share the 828x's MIDI OUT port. This means that they share the same set of 16 MIDI channels, too, so try to do this with devices that listen to only one MIDI channel (such as effects modules), which makes it easier to avoid MIDI channel conflicts.

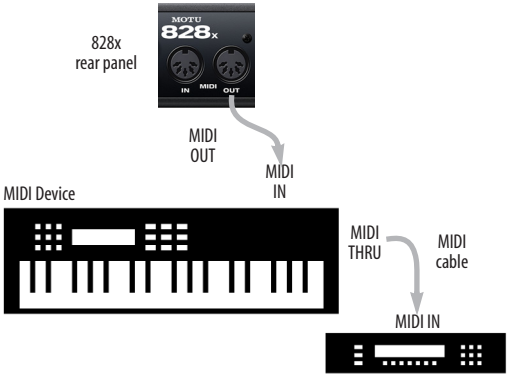


Figure 4-9: Connecting additional devices with MIDI THRU ports.

CONNECT A FOOT SWITCH

If you would like to use a foot switch with your 828x, connect it to the PUNCH IN/OUT jack. See "Quick Reference: MOTU Audio Setup" on page 9 for information about how to program the foot switch to trigger any computer keystroke you wish.

A TYPICAL 828X SETUP

Here is a typical 828x studio setup. This rig can be operated without an external mixer. All mixing and processing can be done in the computer with audio software. During recording, you can use the 828x's CueMix FX no-latency monitoring to listen to what you are recording via the main outs, headphone outs, or any other output pair. You can control monitoring either from the front panel or from the included CueMix FX software. The two front-panel guitar/mic inputs can be routed to outboard effects processors, such as a compressor, EQ or reverb, via the rear panel sends.

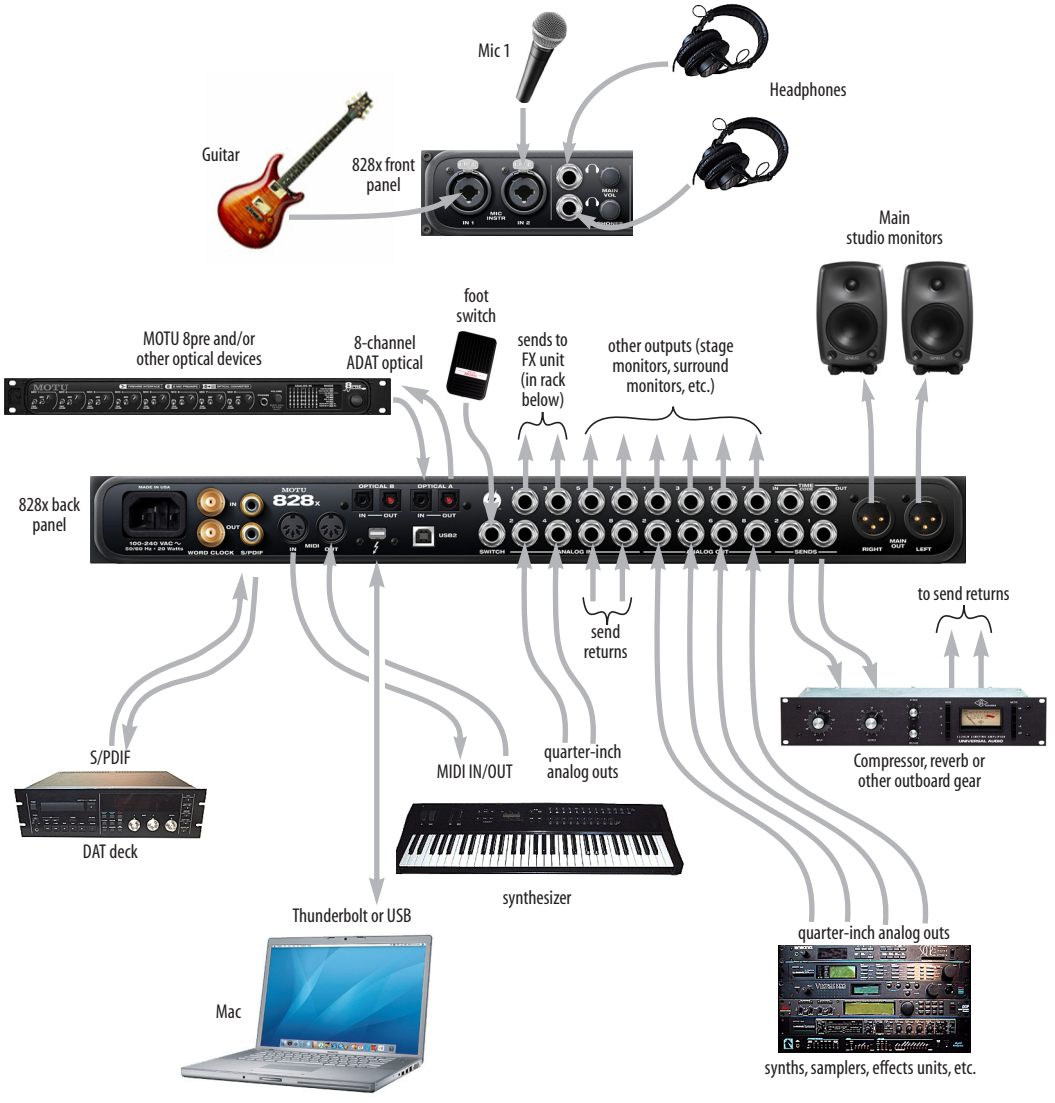


Figure 4-10: A typical 828x studio setup.

OPERATING THE 828X AS A CONVERTER

As explained earlier in "Using optical I/O to operate the 828x as a 16-channel expander" on page 27, the 828x 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 828x optical output to the optical input on another MOTU audio interface, such as a Traveler, 896mk3 or even another 828x. The 828x then serves as a multi-channel expander that adds additional mic, analog TRS and digital inputs to the interface. The benefit of connecting the 828x in this manner (instead of as another interface) is that you can seamlessly integrate the 828x's inputs into the on-board no-latency CueMix monitor mixing in the interface, since the 828x's inputs are fed into CueMix via the interface's optical inputs.

If the device to which you are connecting the 828x 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 27.

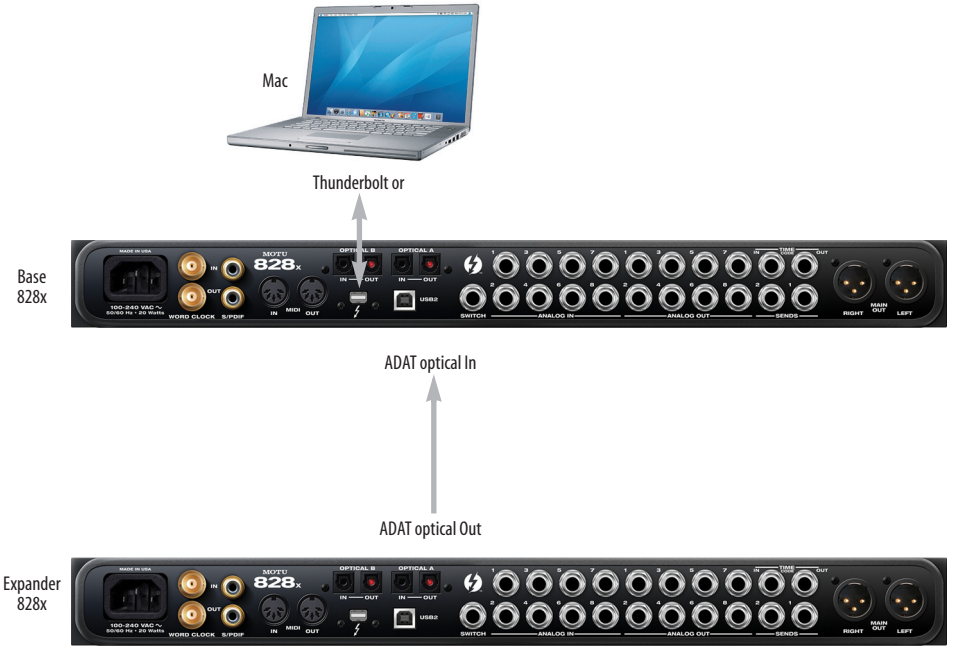


Figure 4-11: Using the 828x as an optical expander. In this example, it is connected to another 828x.

MAKING SYNC CONNECTIONS

If you connect devices digitally to the 828x, or if you need to synchronize the 828x with an outside time reference such as SMPTE timecode, 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 828x?

If you will be using only the 828x's analog inputs and outputs (and none of its digital I/O), and you have no plans to synchronize your 828x system to SMPTE timecode, you don't need to make any sync connections. You can skip this section and proceed to "MOTU Audio Setup" on page 39, where you'll open MOTU Audio Setup to confirm that the *Clock Source* setting is *Internal* as shown below.

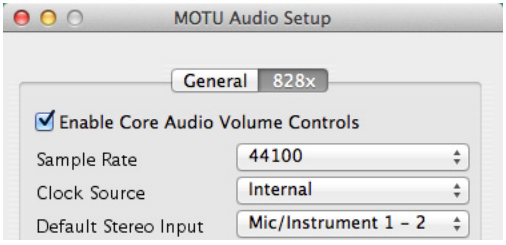


Figure 4-12: You can run the 828x under its own internal clock when it has no digital audio connections and you are not synchronizing the 828x system to an external time reference such as timecode.

Situations that require synchronization

There are three general cases in which you will need to resolve the 828x with other devices:

- Synchronizing the 828x with other digital audio devices so that their digital audio clocks are *phase-locked* (as shown in Figure 4-13)
- Resolving the 828x system to SMPTE timecode 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 828x'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.

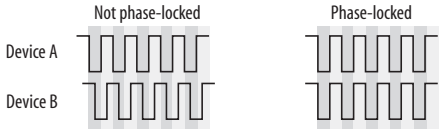


Figure 4-13: When transferring audio, two devices must have phase-locked audio clocks to prevent clicks, pops or other artifacts.

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.

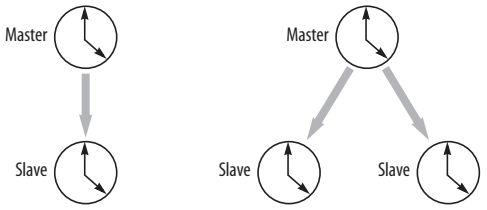


Figure 4-14: To keep the 828x phased-locked with other digital audio devices connected to it, choose a clock master.

Also remember that audio phase lock can be achieved independently of timecode (location). For example, one device can be the timecode 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 828x.

SYNCING TO SMPTE TIMECODE

The 828x system can resolve directly to SMPTE timecode. It can also generate timecode and word clock, under its own clock or while slaving to timecode. Therefore, the 828x 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 828x to slave your audio software to SMPTE as well, via sample-accurate sync (if your host software supports it) or via MIDI Time Code (if your host software supports it).

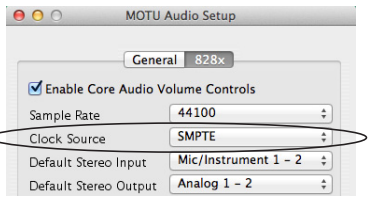
Use this setup if you have:

- ✓ A SMPTE timecode source, such as a multitrack tape deck.
- ✓ An 828x by itself, OR with another slaved device (such as a digital mixer).
- ✓ Host software that supports sample-accurate sync (such as Digital Performer) or MIDI Time Code (such as Pro Tools).

This setup provides:

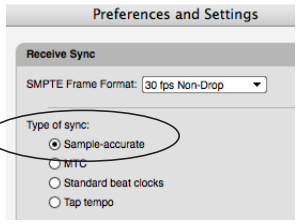
- ✓ Continuous sync to SMPTE timecode.
- ✓ Sub-frame timing accuracy.
- ✓ Transport control from the SMPTE timecode source.

First, choose *SMPTE* as the clock source in AudioDesk, Digital Performer, or MOTU Audio Setup. This setting can also be made in the MOTU SMPTE Setup (shown below).



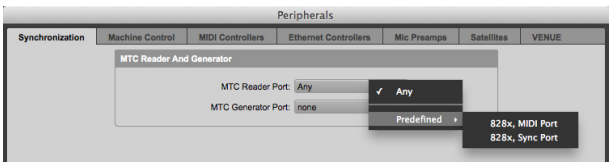
In AudioDesk or Digital Performer:

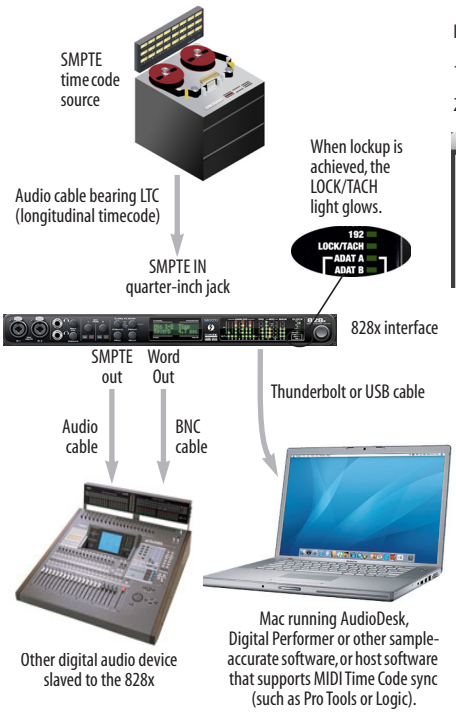
1. Choose *Receive Sync* from the Setup menu.
2. Choose the *Sample-accurate* option. (If this option is grayed out, choose *SMPTE* as the clock source setting first, as shown above.)
3. Make sure that *Slave to External Sync* mode is enabled.



In Pro Tools:

1. Choose *Peripherals* from the Setup menu.
2. Click the Synchronization tab and choose the 828x Sync Port from the MTC Reader Port menu.





Launch the MOTU SMPTE Setup to specify the timecode frame rate and amount of freewheel. Also, confirm that the Clock Source/Address is SMPTE/SMPTE. For details about the other settings, see chapter 10, "MOTU SMPTE Setup" (page 115).

Figure 4-15: Connections for synchronizing the 828x directly to SMPTE timecode.

SYNCING S/PDIF DEVICES

S/PDIF devices will sync to the 828x 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 828x via the S/PDIF connectors. When the device records S/PDIF audio (from the 828x), 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 828x, you'll have to slave the 828x to its S/PDIF input. If you have other digital audio devices connected to the 828x, and they are not slaved directly to the 828x 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 828x via their word clock connection. You can then freely transfer audio between the 828x and the S/PDIF device.

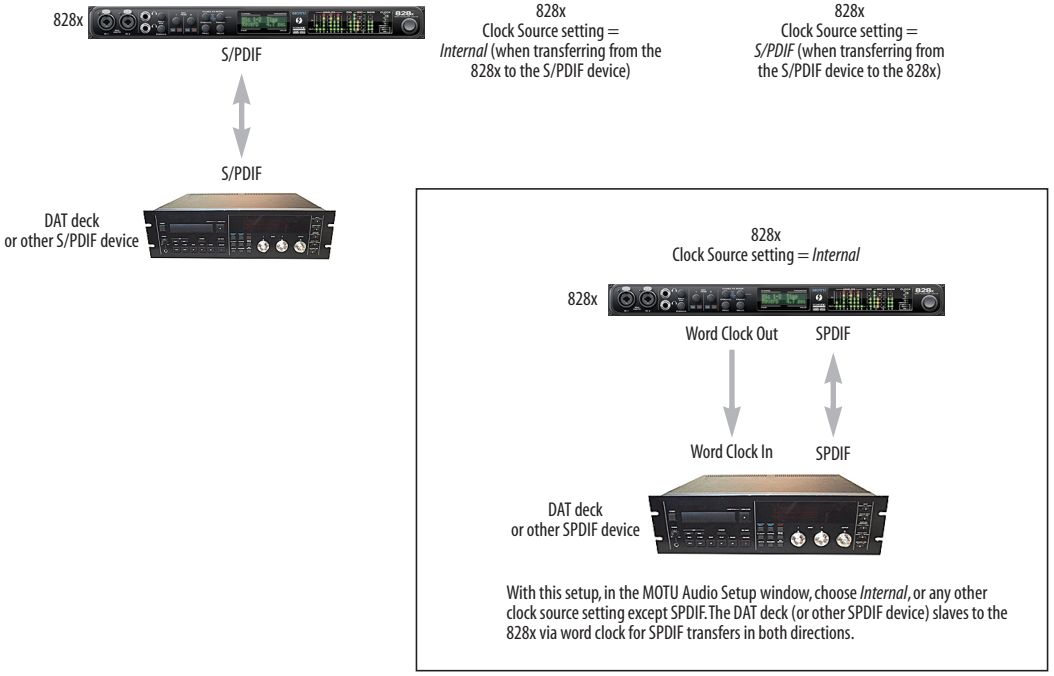


Figure 4-16: Two setups for synchronizing an S/PDIF device with the 828x. In the top diagram, sync is achieved via the S/PDIF connection itself. In this case, you have to choose S/PDIF as the 828x's clock source when recording from the S/PDIF device. If you don't want to have to worry about switching 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 828x or vice versa (not shown). The Word Clock connection maintains sync, regardless of the direction of the transfer.

SYNCING WORD CLOCK DEVICES

The 828x 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 32). 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 4-17 and Figure 4-18.

Figure 4-17: Slaving another digital audio device to the 828x via word clock. For the 828x clock source, choose any source besides word clock, as it is not advisable to chain word clock.

Figure 4-18: Slaving the 828x to word clock. For the 828x 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 word clock distribution device of some kind.

Forcing a 1x word out rate

The 828x 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 828x 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 44.

CONNECTING MULTIPLE 828X INTERFACES

It is possible to connect two or more 828x interfaces to your host computer, and you can mix and match the 828x with other audio interfaces. All connected interfaces will appear as audio I/O devices in any host audio software that supports multiple devices, such as Digital Performer.

Connecting multiple 828x interfaces using Thunderbolt

If your host computer has two or more Thunderbolt ports, you can connect one 828x interface to each port, either directly or at the end of a Thunderbolt daisy chain (on either port or both ports).

Connecting multiple 828x interfaces using Thunderbolt and USB

Another way to operate multiple 828x interfaces on the same host computer is to connect the first 828x to a Thunderbolt port and the second to a USB 2.0 (or 3.0) port.

Multiple interfaces and USB

When connected through USB, the 828x operates as a USB 2.0 device, even when connected to a USB 3.0 port. USB 2.0 provides enough bus speed for several USB 2.0 devices, but due to the 828x's high-performance requirements for real-time operation and low latency timing, we recommend the following:

- Do not connect more than two 828x units to your computer's USB 2.0 or 3.0 port(s).
- When two 828x units are connected with USB 2.0, don't connect other USB 2.0 or 3. 0 devices, such as external hard drives.
- For best results, establish word clock synchronization with other interfaces connected to the same computer.

Mixing and matching audio interfaces

In the scenarios just discussed for connecting multiple 828x interfaces, you can mix and match the 828x with other MOTU audio interface models. For example, you could connect an 828x through Thunderbolt and another MOTU audio interface through USB 2.0.

Multiple interfaces in MOTU Audio Setup

When multiple 828x interfaces are connected, MOTU Audio Setup displays the settings for one interface at a time. To view the settings for an interface, click its tab.

Synchronizing multiple interfaces

If you have multiple 828x interfaces, choose one as the master, and set its Clock Source to *Internal*. Then, click the tab for each of the other 828x units and set their Clock Source to the first 828x.

Resolving to other Core audio drivers

The MOTU Thunderbolt driver can resolve to other Core Audio drivers. This allows the 828x to resolve to other audio interfaces running simultaneously on the same computer, such as the Mac's built-in audio, a MOTU USB audio interface, or even 3rd-party interfaces, without the need for external word clock connections between the devices. Doing so ensures that audio tracks being played or recorded by your 828x interface will not drift apart from tracks on the other device during long playback or recording passes.

To synchronize multiple devices, choose one device (USB or otherwise) as the master clock and then slave the 828x and other devices to it. All interfaces will remain resolved to each other via the master interface.

Make the Clock Source settings for each interface as follows:

- For the master interface, click its tab in MOTU Audio Setup and choose any clock source you wish (except any of the slave interfaces, of course).
- For each slave interface, click its tab and choose the master interface from the *Clock Source* menu. This causes the slave interfaces to resolve to the master interface.

Resolving to word clock

If you need to resolve two 828x interfaces to an external word clock source, resolve the first 828x unit to the external word clock source and then slave the second unit to the first, as shown in Figure 4-18 on page 35. For three interfaces, you can probably get away with daisy-chaining them via their word clock connectors, but this may not always produce the most reliable results. For three or more devices, a word clock distribution device is highly recommended. In this case, the distribution box is the master, and all interfaces slave directly to it (instead of to each other).

Part 2
Using the 828x

CHAPTER 5 MOTU Audio Setup

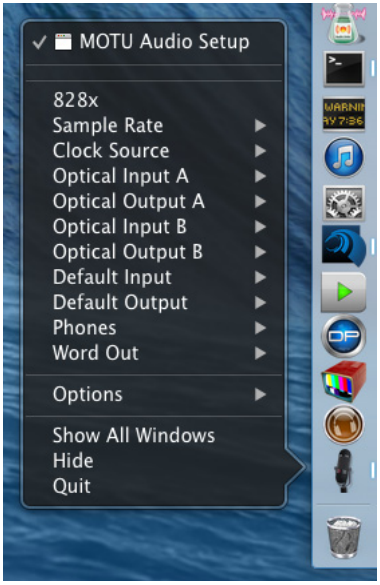
OVERVIEW

MOTU Audio Setup gives you access to basic 828x hardware settings, such as sample rate, clock source, optical format and more.

ACCESSING THE 828X SETTINGS

There are several ways to access 828x settings:

- Open MOTU Audio Setup from the Applications folder.
- Click the MOTU Audio Setup icon in the dock to open the application's window.
- Right-click on the MOTU Audio Setup dock icon to open the shortcut menu shown below.



- From within your host audio software's audio driver preferences. For example, in Digital Performer or AudioDesk, choose *Setup menu> Configure Audio System> Configure Hardware Driver.*

☛ Most applications provide access only to basic settings such as sample rate and clock source. For access to all settings, open MOTU Audio Setup using one of the techniques above.

828x tab settings

The 828x tab (Figure 5-1) provides settings that apply to a specific 828x interface. If you have several 828x (or other MOTU) interfaces connected, you'll see a separate tab for each one.

General tab settings

The General tab (Figure 5-1) provides settings that apply to all connected MOTU audio interfaces.

'828X' TAB SETTINGS

Enable Core Audio Volume Controls

Under Mac OS X 10.6 (Snow Leopard) or later, enabling this option allows you to use the Mac's volume control, such as the Output volume slider shown in Figure 5-2 on page 43, or the volume keys on your Mac's keyboard, to control the output level of the 828x when it is being used as the audio output hardware for your Mac. The *Default Stereo Output* setting (page 42) determines the stereo outputs to be controlled by the Mac's volume controls.

Sample Rate

Choose the desired *Sample Rate* for recording and playback. The 828x 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 828x's optical connectors match the 828x's sample rate. At the 4x sample rates (176.4 or 192kHz), optical I/O 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 Setup.

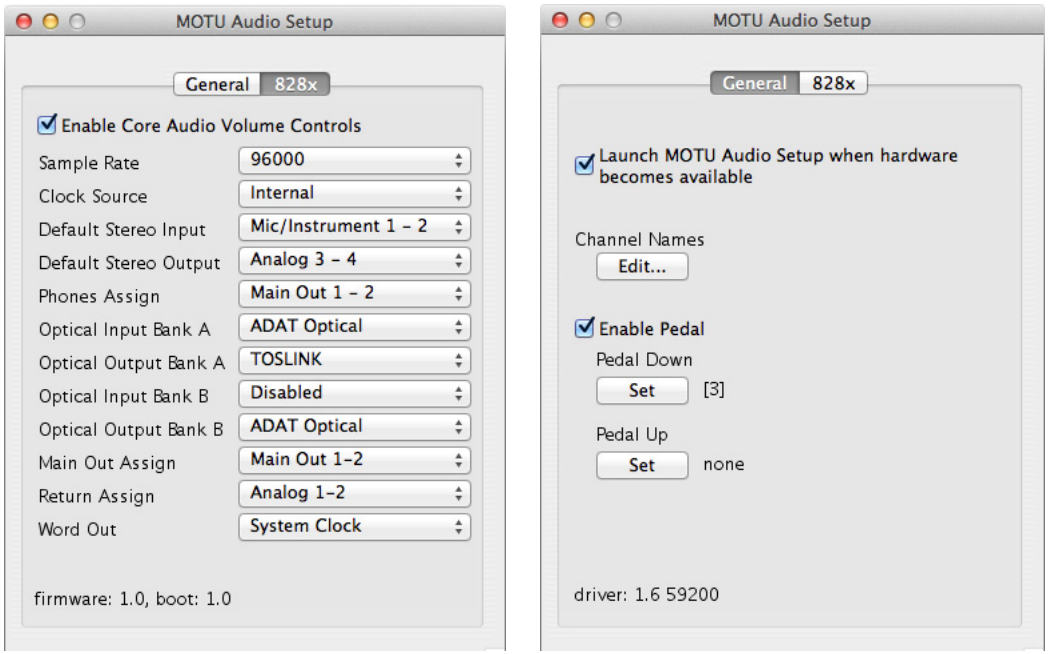


Figure 5-1: MOTU Audio Setup gives you access to all of the settings in the 828x hardware.

Operation at 4x sample rates (176.4 or 192kHz)
At the 4x sample rates (176.4 or 192kHz), operation of the 828x is restricted, due to the higher audio bandwidth demands, as follows:

- All digital I/O is disabled (there is no ADAT optical, TOSLink or S/PDIF input/output).
- The 828x provides 8 channels of analog input and 8 channels of analog output, simultaneously.
- The *stereo return* bus, as described in "Return Assign" on page 43, can only be assigned to one of the four available analog output pairs.
- The headphone output can only be assigned to one of the four available analog output pairs.
- The Main Outs can only be assigned to one of the four available analog output pairs.

Clock Source

The *Clock Source* determines the digital audio clock that the 828x will use as its time base. For a complete explanation of synchronization issues, see "Making sync connections" on page 32. The following sections briefly discuss each clock source setting.

Internal
Use the *Internal* setting when you want the 828x 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 828x system on its internal clock, and then slave the DAT deck to the 828x via the S/PDIF connection (usually DAT decks slave to their S/DIF input when you choose the S/PDIF input as their record source) or via the 828x'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 32.

Word Clock In
The *Word Clock In* setting refers to the Word Clock In BNC connector on the 828x rear panel. Choosing this setting allows the 828x to slave to an external word clock source, such as the word clock output from a digital mixer or another 828x.

S/PDIF
The *S/PDIF* clock source setting refers to the S/PDIF RCA input jack on the 828x. This setting allows the 828x 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 828x. It is not necessary in the opposite direction (when you are transferring from the 828x to the DAT machine).

For further details about this setting, see "Syncing S/PDIF devices" on page 34.

ADAT Optical A / B
The *ADAT optical* clock source settings (*ADAT Optical A* and *ADAT Optical B*) refer to the clock provided by the 828x's two optical inputs, when either one is connected to another optical device. These two settings only appear in the 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 43.

This setting can be used to resolve the 828x 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 828x or an external synchronizer. In

this case, the *ADAT Optical* clock source setting lets you slave the 828x to the device itself via its digital input to the 828x.

This setting is also useful if you just need to make a simple, click-free digital transfer between the 828x 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 "Choosing a clock source for optical connections" on page 28.

TOSLink A / B
The *TOSLink* clock source settings (*TOSLink A* and *TOSLink B*) refer to the clock provided by the 828x's two optical inputs, when either one is connected to another optical device. These two settings only appear in the 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 43.

The *TOSLink* clock source setting refers to the clock provided by an optical S/PDIF device connected to the 828x's optical input. This setting can be used to slave the 828x 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 828x or an external synchronizer. In this case, the *TOSLink* clock source setting lets you slave the 828x to the other device via the 828x's optical input.

This setting is also useful if you just need to make a simple, click-free digital transfer between the 828x 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 "Choosing a clock source for optical connections" on page 28.

SMPTE
Choose this setting to resolve the 828x directly to SMPTE time code (LTC) being received via one of the 828x's audio inputs. For details, see "Setting up for SMPTE time code sync" on page 117 and chapter 10, "MOTU SMPTE Setup" (page 115).

Built-in Audio
Choose this setting to resolve the 828x to your Mac's built-in audio. Doing so will ensure that audio streams playing back from or recorded by the 828x will not drift apart from audio streams simultaneously played or recorded by the Mac's built-in mic, speakers or audio output.

Other audio devices (drivers)
The MOTU Thunderbolt Audio Driver has the ability to resolve to other Core Audio drivers. Doing so will ensure that audio streams playing back from or recorded by the 828x will not drift apart from audio streams simultaneously played or recorded by the other devices.

Default Stereo Input/Output

In the System Preferences window (Figure 5-2), Mac OS X lets you choose third-party hardware such as the 828x for your Mac sound input and output. The system input and output can be used for alert sounds and general audio I/O for applications like iTunes, iMovie, etc.

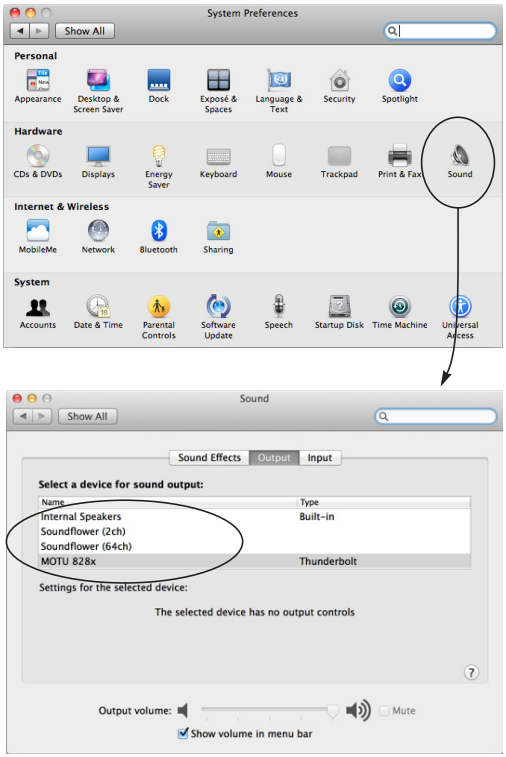


Figure 5-2: The Mac OS X sound preferences let you use the 828x for general stereo audio input and output for your Mac.

The *Default Stereo Input* and *Default Stereo Output* settings in MOTU Audio Setup (Figure 5-1 on page 40) let you specify the stereo input and output on the 828x to be used when it is chosen as the audio I/O device in the system preferences.

Under Mac OS X 10.6 (Snow Leopard) or later, you can use the Mac's volume control, such as the Output volume slider shown in Figure 5-2, or the volume keys on your Mac's keyboard, to control the output level.

Phones Assign

The *Phones Assign* setting lets you choose what you will hear from the headphone jack. Choose *Main Out 1-2* if you'd like the headphone output to match the main outs. 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 8-channel ADAT optical ('lightpipe') and stereo S/PDIF optical ('TOSLINK') as the format for the 828x's optical input and output. 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 input (for a digital mixer) and optical S/PDIF on the 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 828x 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 828x appears in your host software alongside all other 828x inputs, wherever your host software lists them.

The 828x 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 828x 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 828x hardware via CueMix FX (with or without CueMix effects processing on the live inputs).

Word Out

The *Word Out* menu appears when the 828x 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 828x were operating at 176.4kHz, choosing the *Force 44.1/48kHz* option would produce word clock output at 44.1kHz.

'GENERAL' TAB SETTINGS

Launch MOTU Audio Setup when hardware becomes available

Check this option if you would like the MOTU Audio Setup icon to appear in the application dock as soon as a MOTU interface is detected (powered on, plugged in, etc.)

Edit Channel Names

Click the *Edit Channel Names* button to open the Channel Names window (Figure 5-3). This window lets you edit the names of the 828x inputs and outputs, as they appear in your host audio software. For example, when you click on a menu that displays the 828x inputs (or outputs), you will see the names you specify in this window (e.g. "vocal mic", "lead guitar", etc.), instead of the default generic names ("Analog 1", "Analog 2", etc.)

☛ Not all Mac OS X audio software supports channel names. If not, you'll see generic port names in your host audio software. See Appendix B, "Audio I/O reference" page (123).

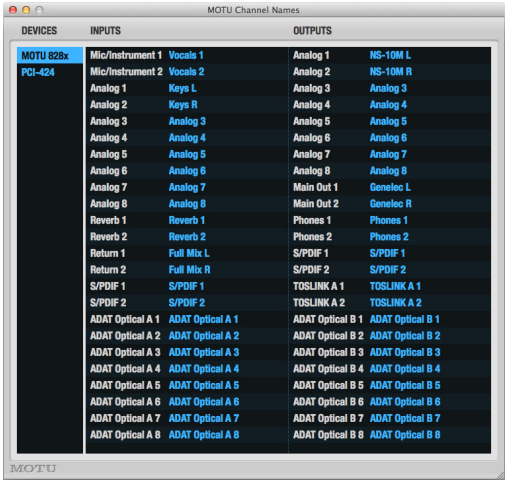


Figure 5-3: The Edit Channel Names window.

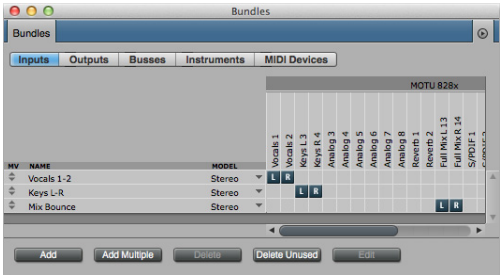


Figure 5-4: 828x channel names as they appear in Digital Performer.

Enable Pedal

Check the Enable Pedal option if a foot switch is connected to the 828x 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.)

CHAPTER 6 Front Panel Operation

OVERVIEW

The 828x offers complete front-panel programming via six rotary encoders and a 2x16 backlit LCD display. All 828x settings can be accessed via these front-panel controls.

MIC/GUITAR INPUTS

The 828x front panel mic/guitar inputs are equipped with many features to handle a wide variety of recording situations.

For information about connections and settings, see "Mic/guitar inputs with preamps" on page 25 in the installation chapter.

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

- "The Inputs tab" on page 74
- "The Channel Settings section" on page 78
- "The Channel tab" on page 78
- "The EQ tab" on page 80
- "The Dynamics tab" on page 86

MAIN VOLUME

The phone jack next to the MAIN VOL knob (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 MAIN VOL knob above it controls the main outs and this jack, but MAIN VOL can be programmed to control any combination of outputs. See "The Monitor Group" on page 91 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 MAIN VOL knob will no longer control the volume of the phone jack, either.

Figure 6-1: The 828x front panel mic/guitar inputs and phone jacks.

PHONES

From the factory, the PHONES jack (Figure 6-1) 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.

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

As explained in the previous section, the LCD provides detailed feedback as you turn the knob (Figure 6-4). To view the current setting without changing it, just push the knob (without turning it).

If you would like the MAIN VOL phone jack and the PHONES jack to output the same signal, assign the PHONES output to mirror the main outs.

METERS AND STATUS LEDS

The meters and LEDs (Figure 6-2) provide complete status and metering information for all 828x inputs and outputs.

Mic/guitar inputs meters with V-Limit™ compressor

The two ten-segment meters for the mic/guitar inputs show input levels from -42 to -1 in the first column of LEDs, plus an additional range in a second column from zero to +12 dB (plus clip). Both 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 turned on, 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. For further information about V-Limit, as well as another feature called *Soft Clip*, see "Overload protection (mic/guitar inputs only)" on page 79.

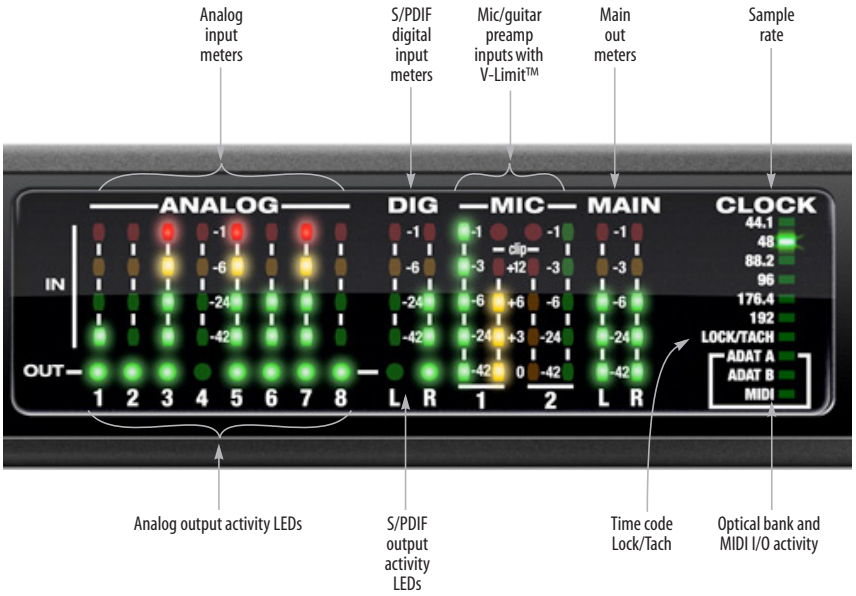


Figure 6-2: The 828x front panel meters and LEDs.

PUSH-BUTTON ROTARY ENCODERS

All of the knobs shown in Figure 6-3 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 LCD provides access to the many 828x 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 an 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. After a brief time-out, the display returns to its previous state before you turned the volume knob.

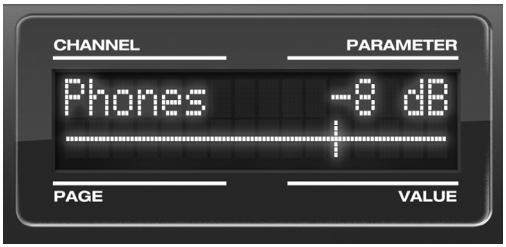


Figure 6-4: The LCD provides feedback as you adjust volume.

Three global menus

Push the PARAM knob to cycle the LCD among three global menus, described in the rest of this chapter:

- 828x SETUP
- AUDIO (settings)
- CUEMIX (mixer)

Figure 6-3: The 828x front panel controls.

828X SETUP MENU

To access the 828x SETUP menu, push the PARAM knob until you see *828x SETUP* displayed in the LCD. This menu provides basic features for managing the 828x hardware. Turn the PARAM knob to access each setting, explained briefly below.

LCD Contrast

Turn the VALUE knob to adjust the LCD contrast.

Figure 6-5: 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 828x 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 the push again to confirm the save. To cancel the save operation at any time, turn the PARAM knob.

Load Preset

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

All Notes Off

The *All Notes Off* setting sends a MIDI *All Notes Off* message, as well as a MIDI note-off message for every note on every MIDI channel. This stops any stuck notes that are currently playing. Push VALUE to send the All Notes Off MIDI data.

Factory Defaults

Push VALUE to restore the 828x 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 Setup software, as shown below:

Audio menu item	Where to find more information
Clock Source*	"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 43
Return Assign	"Return Assign" on page 43
Optical In/Out A/B*	"Optical input/output" on page 43
ADAT SMUX Type	See below. This menu item is only available when the 828x is operating at 88.2 or 96 kHz.
Word Clock Out	"Word Out" on page 44

** If the 828x is currently connected to a computer, this setting cannot be changed from the front-panel LCD. It must be changed in MOTU Audio Setup instead. Or, you can disconnect the 828x from the computer to change the Clock Source from the front panel.*

ADAT SMUX Type

When the 828x 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 828x 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 69) before learning the LCD, especially "CueMix FX basic operation" on page 70.

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 6-6). This is similar to accessing the Inputs tab in CueMix FX ("The Inputs tab" on page 74).

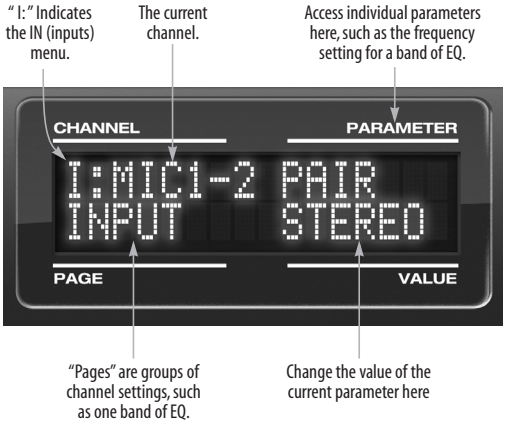


Figure 6-6: The IN (inputs) menu.

Choosing a channel

Once you see the Inputs menu (Figure 6-6) 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 74).

Choosing a setting to modify

Once you've selected an input channel, you can access the various settings for that channel using the PAGE 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 74), as well as the settings in the Channel tab (Figure 9-8 on page 78).

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 80), Dynamics tab (Figure 9-21 on page 86) and Reverb tab (Figure 9-23 on page 89) 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 52.

The OUT (Outputs) menu

Push the CHANNEL button repeatedly until you see "O:" in the channel section of the LCD (Figure 6-7). This is similar to accessing the Outputs tab in CueMix FX ("The Outputs tab" on page 76).

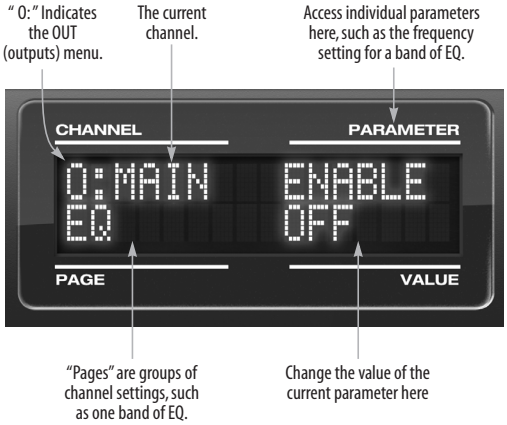


Figure 6-7: The OUT (outputs) menu.

Choosing a channel
Once you see the Outputs menu (Figure 6-7) 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 77).

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 and PARAMETER knobs. The knobs function the same as described for inputs in "Choosing a setting to modify" on page 49 and "Adjusting the value of a parameter" on page 50.

Summary of Output menu settings
For a summary of Inputs menu settings, see "Outputs menu" on page 52.

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 6-8). This is similar to accessing the Mixes tab in CueMix FX ("The Mixes tab" on page 72).

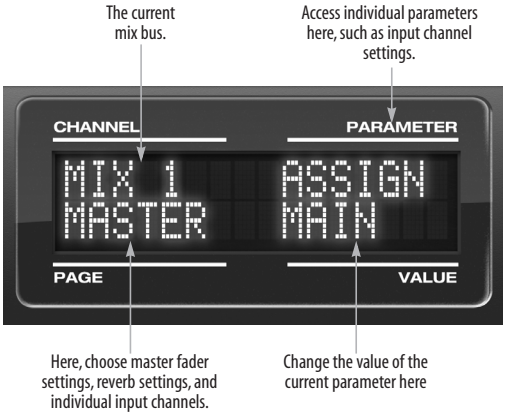


Figure 6-8: The MIX (Mixes) menu.

Choosing a mix bus

Once you see the MIX menu (Figure 6-8) 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 72).

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 and PARAMETER knobs. The knobs function the same as described for inputs in "Choosing a setting to modify" on page 49 and "Adjusting the value of a parameter" on page 50. 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.
- 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 and VALUE knobs.

Summary of Mix menu settings

For a summary of Mix menu settings, see "Mixes menu" on page 53.

The REVERB menu

Push the CHANNEL button repeatedly until you see "REVERB" in the channel section of the LCD (Figure 6-9). This is similar to accessing the Reverb tab in CueMix FX ("The Reverb tab" on page 89).

☛ The REVERB processor is not available at sample rates above 48 kHz. Therefore, when the 828x is operating at 88.2 kHz or higher, the REVERB menu does not appear in the LCD.

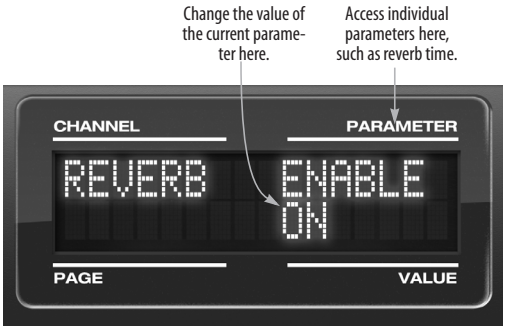


Figure 6-9: The REVERB menu.

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 53.

INPUTS MENU

CHANNEL	PAGE	PARAM
INPUTS mic 1-2 Analog 1-2 Analog 3-4 etc.	INPUT	PAIR PHASE L-R/M-S (stereo or M/S) SWAP WIDTH TRIM PAD PHANTOM LIMITER LOOKAHEAD SOFTCLIP (Mic only)
	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
	MIC 1 MIC 2 (if mono)	MUTE SOLO (BAL/WID) - n/a PAN FADER
	MIC 1-2 (if stereo)	MUTE SOLO BAL/WID WIDTH FADER
	ANALOG 1 ANALOG 2 etc. (if mono)	MUTE SOLO (BAL/WID) - n/a PAN FADER
	ANALOG 3-4 ANALOG 5-6 etc. (if stereo)	MUTE SOLO BAL/WID BAL FADER
	SPDIF	MUTE SOLO BAL/WID BAL FADER
	ADAT A1-2 ADAT A3-4 etc.	MUTE SOLO BALD/WID BAL FADER
	ADAT B1-2 ADAT B3-4 etc.	MUTE SOLO BALD/WID BAL 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 828x's memory, and they remain in effect even when the 828x is not connected to a computer. This allows you to use the 828x 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 828x provides multi-channel audio input and output for Core Audio-compatible audio applications, including MOTU's Digital Performer and AudioDesk, Apple's Logic Pro, Logic Express, SoundTrack Pro, and GarageBand, and other third-party software applications such as Ableton Live, Avid Pro Tools, Cockos Reaper, Propellerhead Reason and Record, Steinberg Cubase and Nuendo, and others.

AudioDesk is included with your 828x system. For complete information about all of AudioDesk's powerful workstation features, refer to the *AudioDesk User Guide* included with your 828x system.

Digital Performer, MOTU's state-of-the-art digital audio workstation software, is available separately; for details about upgrading from AudioDesk to Digital Performer, talk to your authorized MOTU dealer or visit MOTU.com.

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 828x Software" (page 19)

chapter 4, "Installing the 828x Hardware" (page 23).

RUN MOTU AUDIO SETUP

Before you run your host audio software, launch MOTU Audio Setup to configure your 828x hardware.

MOTU Audio Setup 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 Setup" (page 39).

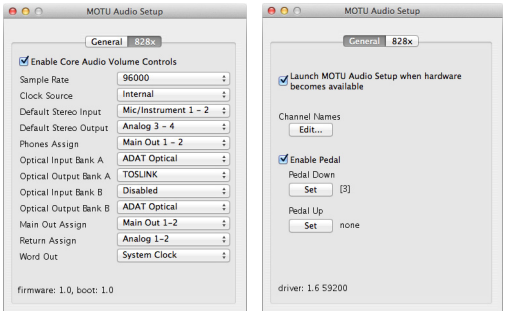


Figure 7-1: MOTU Audio Setup.

The following sections provide a brief explanation of each 828x setting.

Sample rate

Choose the desired overall sample rate for the 828x system and your host audio software. Newly recorded audio will have this sample rate.

Clock Source

This setting is very important because it determines which audio clock the 828x will follow.

If you do not have any digital audio connections to your 828x (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 devices connected to the 828x optical ports, see "Choosing a clock source for optical connections" on page 28.

If you are slaving the 828x and your host software to SMPTE time code, follow the directions in chapter 10, "MOTU SMPTE Setup" (page 115).

Phones Assign

This 828x 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.

Optical input and output

To make a 828x 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 main outs of the 828x. If you would like to treat them as their own separate output pair, choose *Main Out 1-2*.

Return Assign

In your host audio software audio input menus, you'll see an 828x input called *Return 1-2*. This is a stereo feed from the 828x that matches the signal of one of its output pairs. Use the Return Assign menu in MOTU Audio Setup 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 828x 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 output as the Return Assign setting.

Reverb return

The 828x 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 is a menu that lets you choose among various drivers that may be in your system. Choose the MOTU Audio driver from this menu.

Digital Performer and AudioDesk

In MOTU Digital Performer or AudioDesk, choose *Configure Audio System* > *Configure Hardware Driver* from the Setup menu, as shown in Figure 7-2. Choose *MOTU 828x* from the list of CoreAudio drivers.

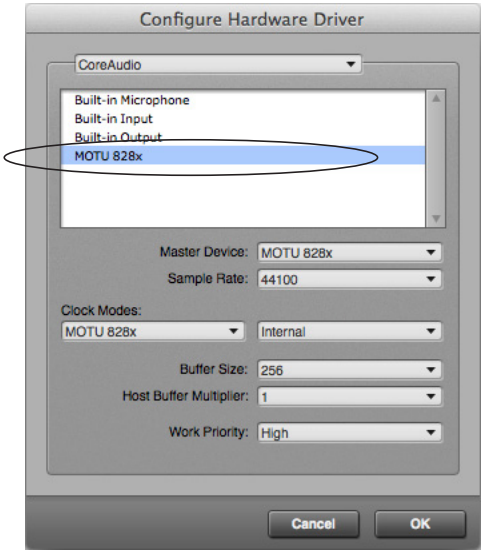


Figure 7-2: Enabling the 828x in Digital Performer

Pro Tools

In Avid Pro Tools, go to the Setup menu and choose *Playback Engine* as shown in Figure 7-3. Choose the *MOTU 828x* from the Current Engine menu. For information about the *H/W Buffer Size* setting, see "Adjusting the audio I/O buffer" on page 65.

☛ Pro Tools 9 or later is required for operation with the 828x.

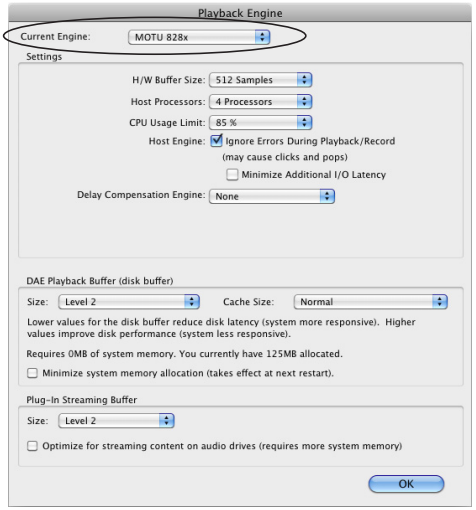


Figure 7-3: Enabling the 828x in Pro Tools

Logic Pro and Logic Express

In Logic Pro and Logic Express, go to the Preferences window, click the *Audio* tab, click the *Devices* tab and click the *Core Audio* tab as shown in Figure 7-4. Choose the *MOTU 828x* from the Output and Input device menus.

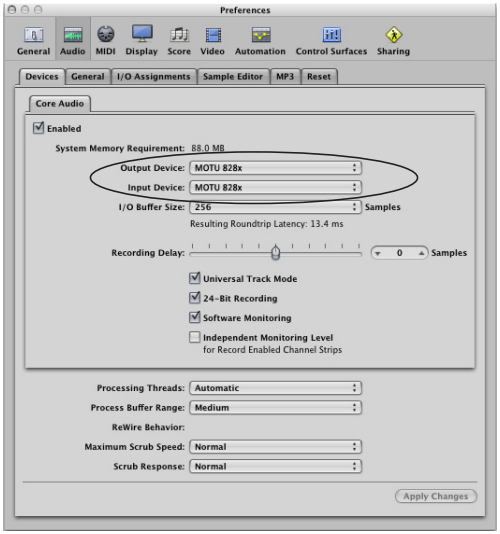


Figure 7-4: Enabling the 828x in Logic

Garage Band

In Garage Band, go to the *Audio/MIDI* preferences and choose *MOTU 828x* from the *Audio Output* and *Audio Input* menus as shown below in Figure 7-5.

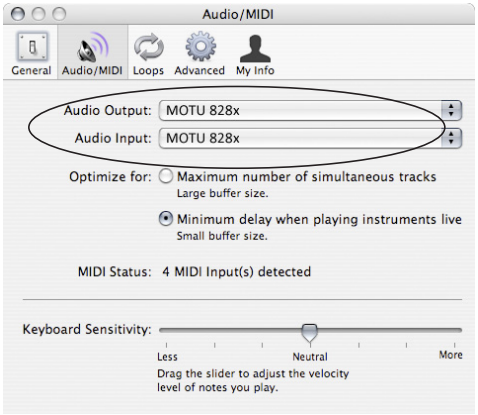


Figure 7-5: Enabling the 828x in Garage Band

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 828x* from the *VST Audio System* menu as shown below in Figure 7-6. Activate the inputs and outputs within Cubase or Nuendo as usual.

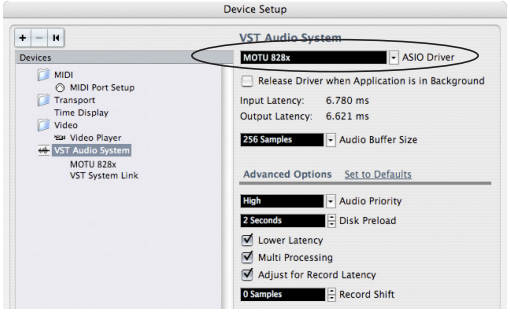


Figure 7-6: Enabling the 828x in Cubase

Live

In Ableton Live, access the preferences window and click the *Audio* tab. Choose *Core Audio* from the *Driver Type* menu. Choose the *MOTU 828x* from the *Input Audio Device* and *Output Audio Device* menus as shown below in Figure 7-7. To enable or disable input or output channels, click the *Input Config* or *Output Config* buttons.

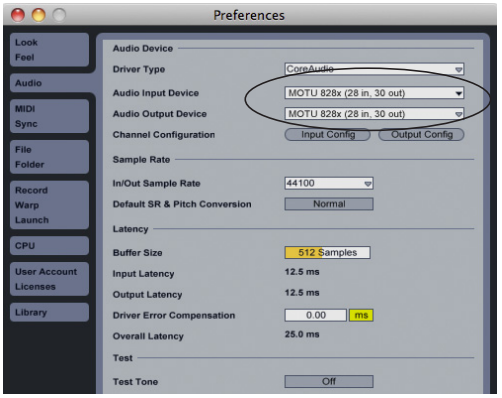


Figure 7-7: Enabling the 828x in Live

Reason and Record

In Propellerhead Reason or Record, go to the Preferences window, choose *Audio* preferences from the menu and choose *MOTU 828x* from the *Audio Output* menu as shown below in Figure 7-8.

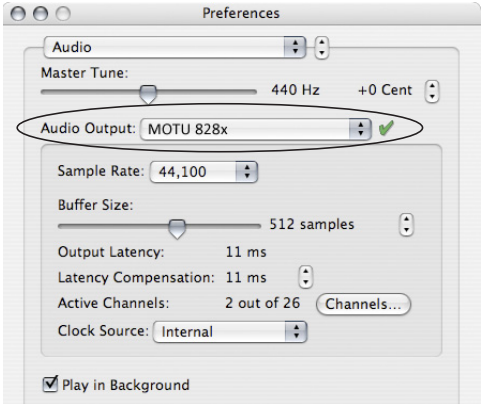


Figure 7-8: Enabling the 828x in Reason

Reaper

In Cockos Reaper, access the Preferences and click *Devices* under the *Audio* preferences. Choose *MOTU 828x* from the Audio Device menu as shown below in Figure 7-9.

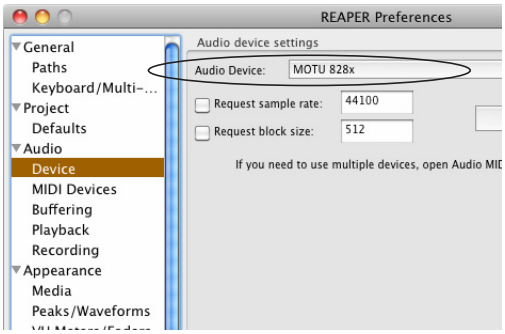


Figure 7-9: Enabling the 828x in Reaper

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 Mac OS X, audio I/O buffer size is handled by the host audio application rather than the 828x Core Audio driver.

Most audio software applications provide 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 63).

WORKING WITH 828X INPUTS AND OUTPUTS

Once you've enabled the 828x's driver, 828x 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 Setup to make sure they are enabled. 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 Setup to mirror other outputs. In addition, the 828x 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 43, "Phones Assign" on page 43 and "Main Out Assign" on page 43.

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 16 inputs and 16 outputs provided by your 828x.

Mix1 1-2 return bus

In your host audio software audio input menus, you'll see an 828x input called *Mix1 Return 1-2*. This is a stereo feed from the 828x that matches its main outs (Mix 1). This can be used, for example, to record a final stereo mix for reference and archiving purposes.

☛ Warning: the Mix1 Return 1-2 return inputs can cause feedback loops! DO NOT assign this input to a track assigned to the 828x main outs.

For Mix1 Return 1-2, you can choose whether or not to include audio being sent to the main outs from the computer. See "Mix1 Return Includes Computer" on page 94.

The "Phones 1-2" output

Host audio tracks assigned to the *Phones 1-2* output pair will be heard on the headphone jack only. If the Phones are set to mirror another output (such as the main outs), the mirrored audio stream is mixed with the phones audio output. For further explanation, see "Phones Assign" on page 43.

Audio Input and output names

The 828x audio driver supplies text string labels for its 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 B, "Audio I/O reference" page (123).

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 63).

WORKING WITH CUEMIX FX MIXING AND EFFECTS

The 828x provides powerful external mixing, EQ, compression and reverb, which you can operate hand-in-hand with your host's mixing environment. For example, the 828x 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 a 828x 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 828x settings as a file in your host project folder for instant recall of all settings. See chapter 9, "CueMix FX" (page 69) for complete details.

SYNCHRONIZATION

You may encounter situations in which you need to synchronize your audio software and the 828x to the other components of your system.

Synchronizing digital audio connections

If you have devices connected to the 828x's digital optical ports, you need to be concerned with the synchronization of the 828x's digital audio clock with other devices connected to it digitally (if any). For example, if you have a digital mixer connected to the 828x 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 28. If you don't have any digital audio devices connected to the 828x, digital audio phase-lock does not apply to you.

Resolving directly to time code (with no synchronizer)

If your host audio software supports Core Audio's sample-accurate positioning protocol, or if it can slave to MIDI Time Code, then it can resolve to the 828x's built-in time code synchronization feature. To resolve your 828x directly to SMPTE time code with no additional synchronization devices, use the setup shown in "Setting up for SMPTE time code sync" on page 117.

USING A FOOT SWITCH

Use a foot switch connected to the 828x to trigger recording punch-in and punch-out, or any other feature in your host audio software that is assigned to a computer keystroke. By default, the foot switch triggers the 3 key on the computer keypad. To trigger a different set of keystrokes with the foot switch, visit MOTU Audio Setup. (See "Enable Pedal" on page 44.)

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 828x input, passes through the 828x hardware into the computer, through your host audio software, and then back out to an 828x 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 828x's CueMix FX digital mixer to patch the input directly to your monitor outs via the 828x audio hardware. The 828x 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 67.

If you *do* need to process a live input with host software plug-ins, or if you are playing virtual instruments live through your 828x 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.

MONITORING LIVE INPUT

There are two ways to monitor live audio input with an 828x: 1) through the computer or 2) via the 828x 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 828x-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 828x Classic Reverb.

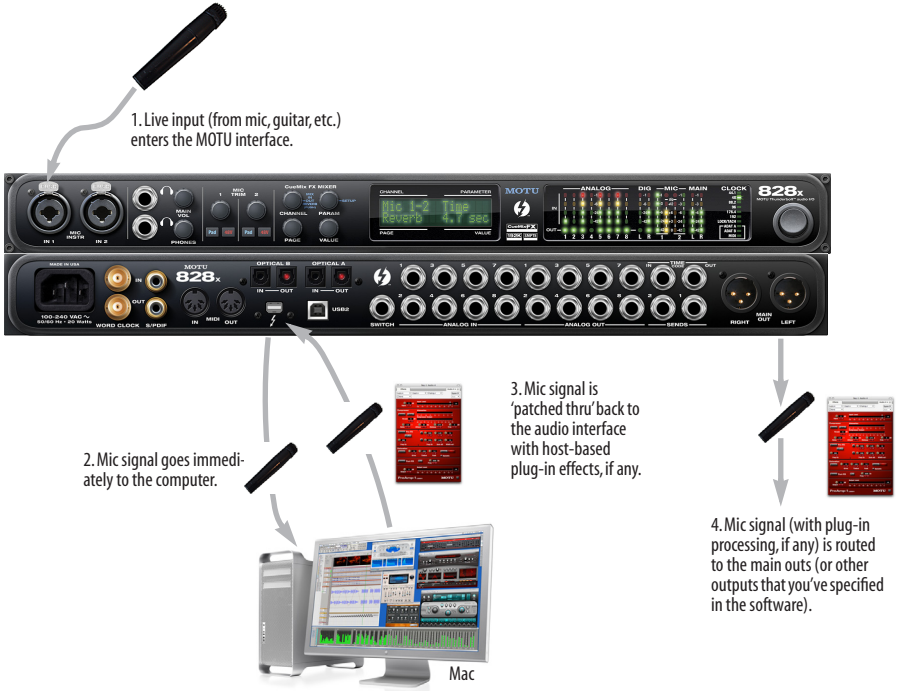


Figure 8-1: There are two ways to monitor live audio inputs with an 828x: 1) through the computer or 2) via CueMix FX hardware monitoring. This diagram shows method 1 (through the computer). When using this method, use your host software's buffer setting to reduce the slight delay you hear when monitoring 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 828x, 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.

Under Mac OS X, audio I/O buffer size is handled by the host audio application (not the 828x Core Audio driver). Most audio software applications provide an adjustable audio buffer setting that lets you control the amount of delay you'll hear when monitoring live inputs or processing them with software plug-ins. Below are a few examples.

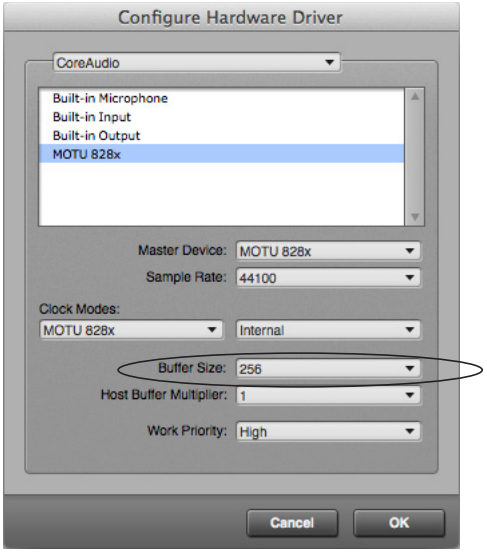


Figure 8-3: In Digital Performer and AudioDesk, choose Setup menu> Configure Audio System> Configure Hardware Driver to open the dialog shown above and access the Buffer Size setting. Refer to your Digital Performer or AudioDesk manual for information about the Host Buffer Multiplier setting.

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 828x effects, such as the reverb, EQ and/or compression.

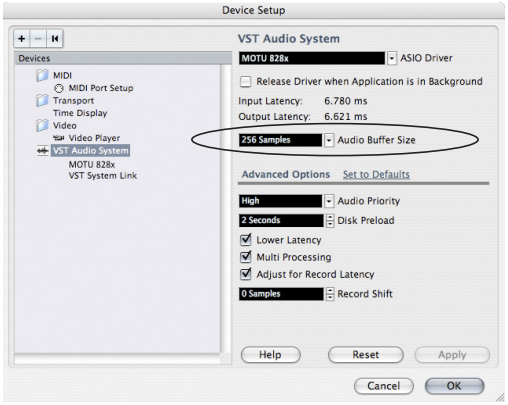


Figure 8-4: In Cubase or Nuendo, choose Devices menu > Device Setup. Select MOTU 828x, then click the Control Panel button to access the window above and the Buffer Size setting.

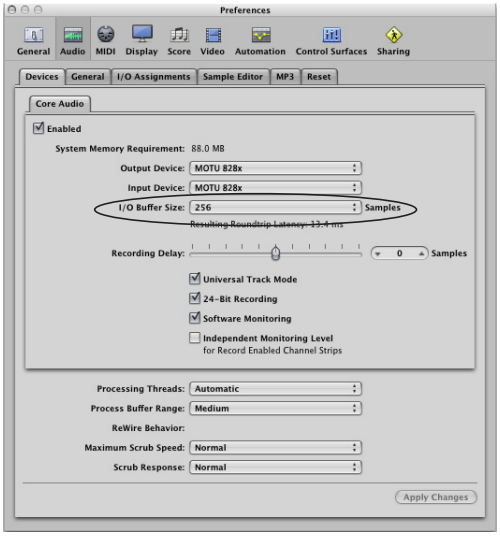


Figure 8-5: In Logic Pro or Logic Express, go to the Audio Driver preferences to access the Buffer Size option shown above.

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 AudioDesk, Digital Performer or other 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.

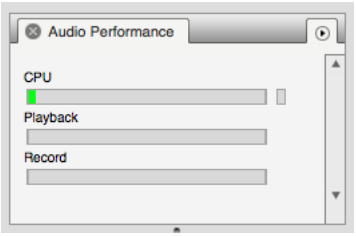


Figure 8-6: When adjusting the buffer size to reduce monitoring latency, watch the 'processor' meter in Digital Performer or AudioDesk's Performance Monitor. If you hear distortion, or if the Performance meter is peaking, try raising the buffer size.

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 828x has a more direct method of patching audio through the system: *CueMix FX*. CueMix FX is a digital mixer in the 828x hardware 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 828x itself and are mixed with audio playback from your host 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 the 828x's CueMix FX software
- From within your host audio software (if it supports direct hardware monitoring)

You can even use both methods simultaneously.

Using the CueMix FX app

If your host audio software does not support direct hardware monitoring, you can 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 828x 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 69).

Controlling CueMix FX from your audio software

Some audio applications 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 828x input directly to an output when you record-arm a track. Exactly how this is handled depends on the application.

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

- Digital Performer
- AudioDesk

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 AudioDesk or Digital Performer

To turn on CueMix FX in AudioDesk and Digital Performer:

1 From the Setup menu, choose *Configure Audio System>Input Monitoring Mode*.

2 Choose the *Direct hardware playthrough* option, as shown below in Figure 8-7.

3 From the Studio menu, choose *Audio Patch Thru*, and choose any monitoring mode except *Off*.

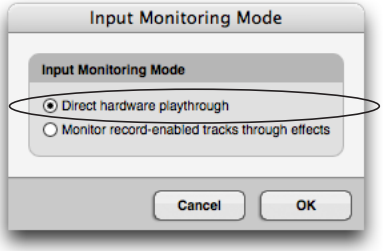


Figure 8-7: Enabling CueMix FX in AudioDesk or Digital Performer.

Once enabled, CueMix FX monitoring is tied with Digital Performer or AudioDesk's *Audio Patch Thru* feature: when you record-enable a track, the track's input is routed directly to its output (via CueMix FX in the 828x hardware). For example, if you record-enable a track called *guitar* in your DP or AudioDesk project, and its audio input assignment is *Analog in 2*, and its audio output assignment is optical channels 7-8, CueMix FX no-latency hardware monitoring will automatically be set up from analog in 2 to optical outputs 7-8.

Controlling CueMix DSP from within other software

Consult the manual for your software.

CHAPTER 9 CueMix FX

OVERVIEW

CueMix FX is a cross-platform software application that provides graphic, on-screen control for the 828x's flexible CueMix FX on-board mixer and effects processing.

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

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

A 16-BUS MIXER WITH EQ, COMPRESSION AND REVERB

All 828x inputs 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 828x hardware, independent of the computer. Effects can even be applied when the 828x is operating stand-alone (without a computer). 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 828x's flexible effects architecture allows you to apply EQ and compression on every input and output (a total of 32 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 828x'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.

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 828x'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 828x to operate as a portable, stand-alone mixer with effects.

OPENING CUEMIX FX

CueMix FX is an application installed with the rest of your 828x software installer. It can be found in the Applications folder.

CUEMIX FX BASIC OPERATION

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

Eight stereo mix busses

CueMix 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 828x 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 72). 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.

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 74) gives you access to settings for individual 828x inputs (or input pairs), such as phase, trim, EQ and dynamics processing. Each input also includes a send to the 828x'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 77) gives you access to settings for each 828x output pair, including EQ, dynamics processing and send/return controls for feeding and returning the output signal to/from the 828x'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 828x has a global reverb module (Figure 9-23 on page 89). Once it has been activated, you can feed signals to the reverb processor from various points in the 828x 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, and more.

Widening the CueMix FX window

To view more input faders at once, resize the window.

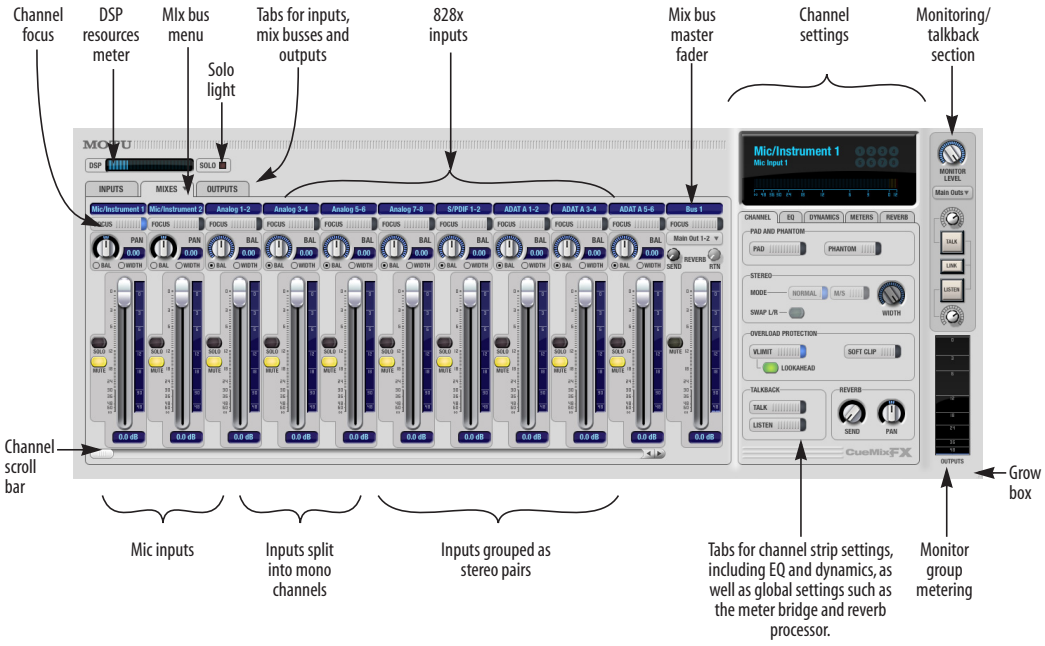


Figure 9-1: CueMix FX is a virtual mixer that gives you control over the 828x's on-board mixing features.

THE MIXES TAB

Click the *Mixes* tab (Figure 9-2) to gain access to the 828x'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 828x output pair to which each bus master fader is assigned. If the optical output bank is either disabled or set to TOSLink (see "Optical input/output" on page 43), then bus 5, 6 and 7 are disabled.

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) 828x 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.

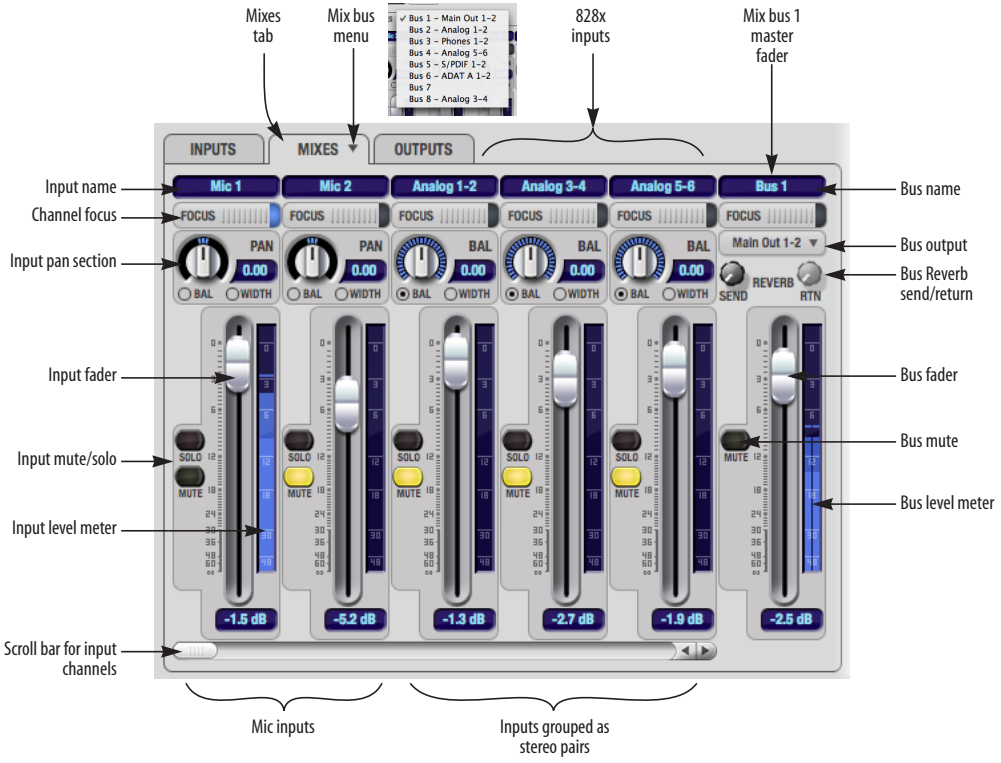


Figure 9-2: The Mixes tab.

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 828x'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 828x'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 90.

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 828x inputs.

Naming an input
Click the input name at the top of the input channel strip (Figure 9-2) to edit the name. Input names are global across all mixes. This name also appears in host audio software on the computer (if the software supports channel names).

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 78). 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 95.

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 828x provides many features for managing analog and digital input signals. Some of these features, such as the 828x'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 828x 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 828x'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: trim, overload protection, phase, stereo versus M/S decoding, width, L/R swap, EQ, dynamics and reverb send.

Input name

Click the input name at the top of the input channel strip (Figure 9-3) to edit the name. Input names are global across all mixes. 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 78).

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 828x 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 828x input:

Input	Trim cut	Trim boost	Trim Range
Mic/Guitar	0 dB	53 dB	53 dB
TRS analog inputs	-96 dB	+22 dB	118 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 94 and "Configurations menu" on page 112.

Input EQ and dynamics

The 828x 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 EQ tab 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 78.

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 80).

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.

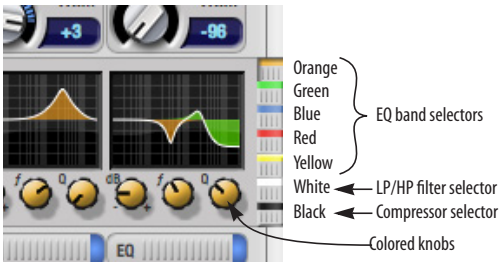


Figure 9-4: The EQ/Dynamics selectors.

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.

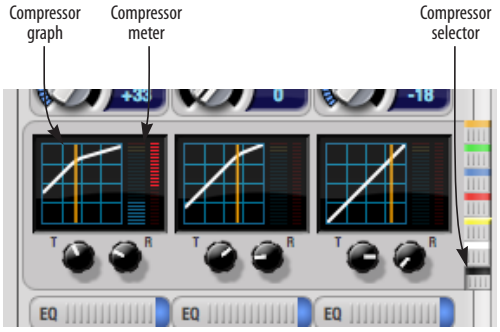


Figure 9-5: The Compressor controls.

☛ 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 80.

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 828x'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 828x. This processing occurs at the very end of the signal flow, after everything else (host based effects, 828x 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

Click the output name at the top of the output channel strip (Figure 9-6) to edit the name. 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 78). 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 95.

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 75.

Output reverb send/return

The output reverb *send* (Figure 9-6) feeds the signal for the output to the 828x'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 *Mixes* to eliminate the possibility for feedback loops created by reverb send/return loops. See "Split point" on page 90. It is also disabled when the 828x is operating at the 88.2 kHz sample rate or above, as the reverb is not available at high sample rates.

The output reverb *return* (Figure 9-6) feeds the output of the 828x'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 processing, 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 92.

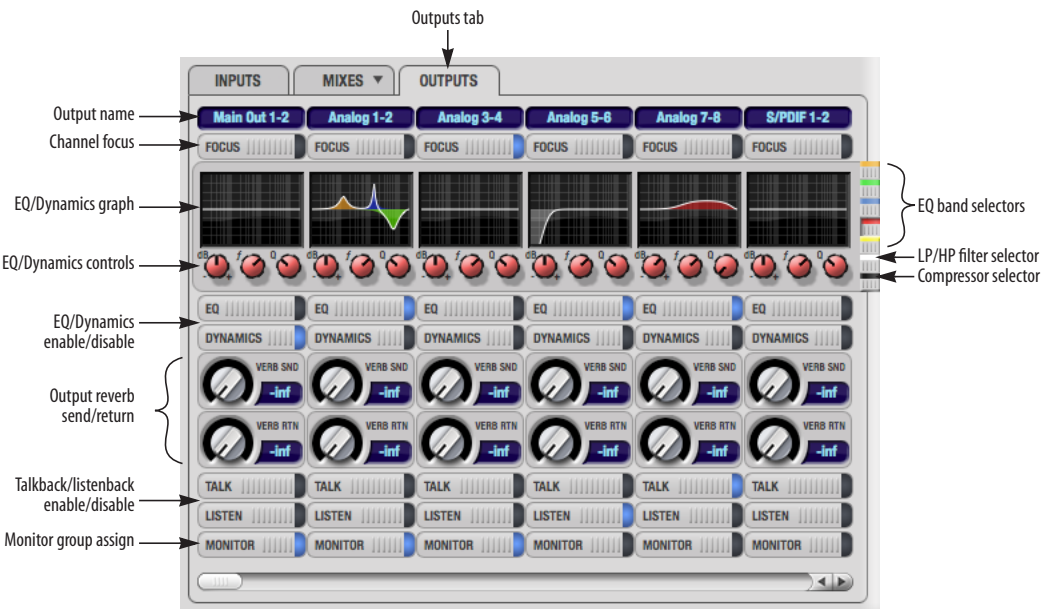


Figure 9-6: The Outputs tab.

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 91.

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.

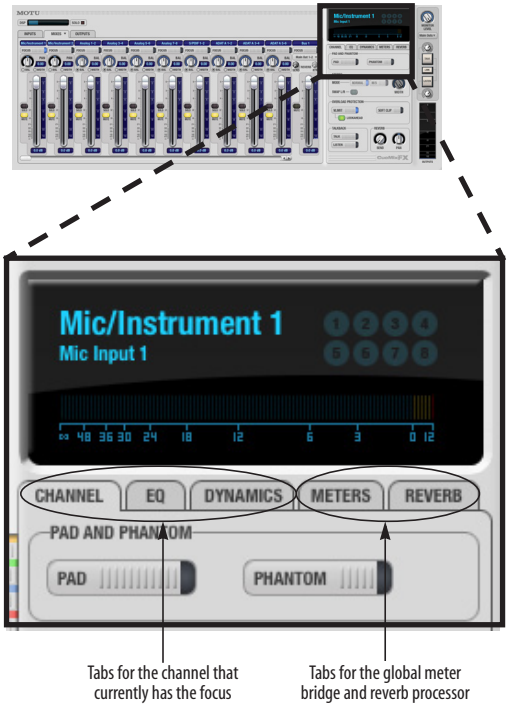


Figure 9-7: The Channel Settings section.

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.

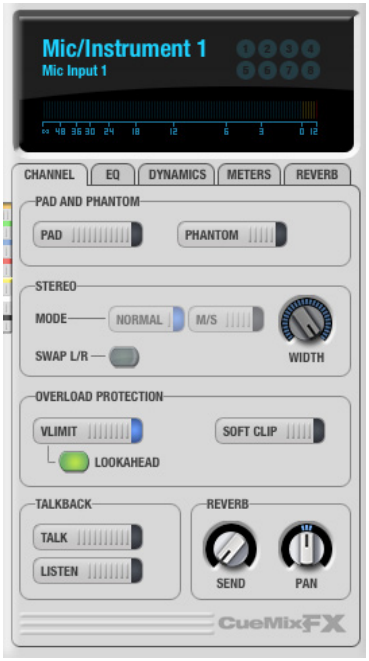


Figure 9-8: The Channel tab.

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 74). 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 828x hardware itself. 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 73.

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

Overload protection (mic/guitar inputs only)
The Overload Protection section (Figure 9-8) provides two features that help prevent digital clipping on the two front-panel mic/guitar inputs. These options are only available on these two preamp-equipped 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.

Additional or alternative protection can be applied to the mic/guitar inputs by enabling *Soft Clip* (Figure 9-8). When enabled, Soft Clip engages just before clipping occurs and helps further reduce perceptible distortion.

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 92.

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 76. 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 89) 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 828x. 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 828x *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 model 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 828x 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 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.

☛ Shortcut: hold down the Option/Alt key while clicking anywhere in the EQ graph that's not directly on an EQ filter handle to enable or bypass the EQ for that input or output.

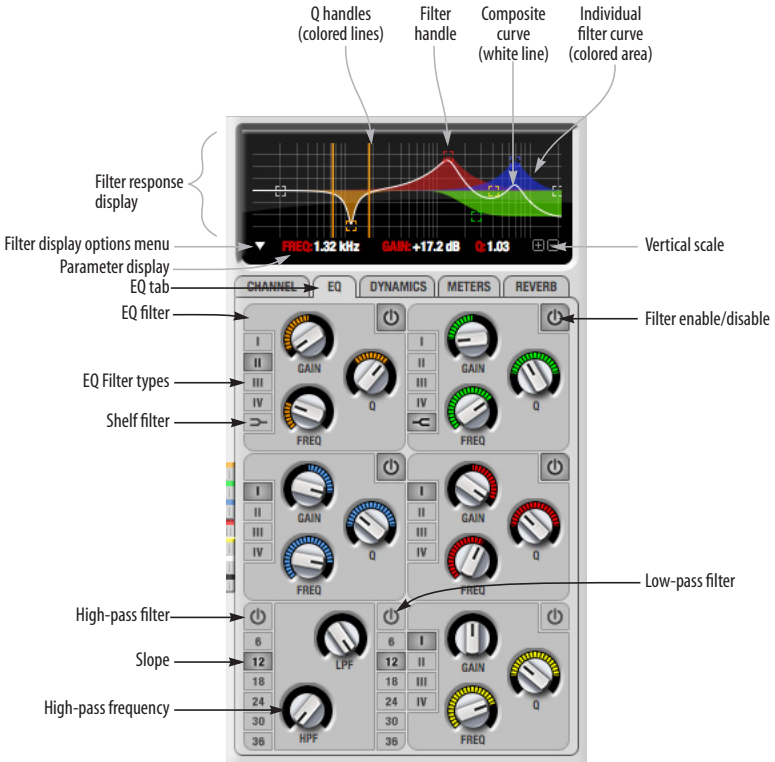


Figure 9-10: The EQ tab.

☛ EQ is disabled when the 828x is operating at a 4x sample rate (176.4 or 192 kHz).

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:

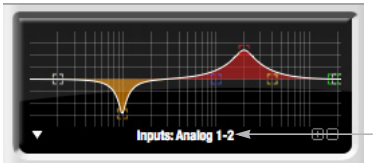


Figure 9-11: When a filter handle is not selected and when the cursor is not hovering over the display, the parameter display shows the name of the current channel being edited (the channel that currently has the focus).

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:

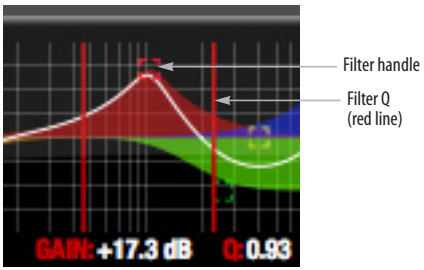


Figure 9-12: Drag the filter handle to adjust its frequency and/or boost/cut. Drag the Filter Q handles to adjust the Q.

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. Hold Shift to adjust gain only. Hold Control to adjust frequency only.

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 = +3.2, +10 and +20 dB

Type I

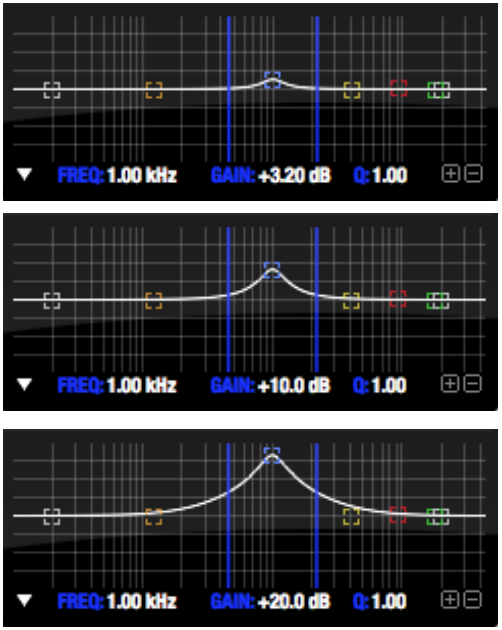


Figure 9-13: Type I EQ filter style.

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

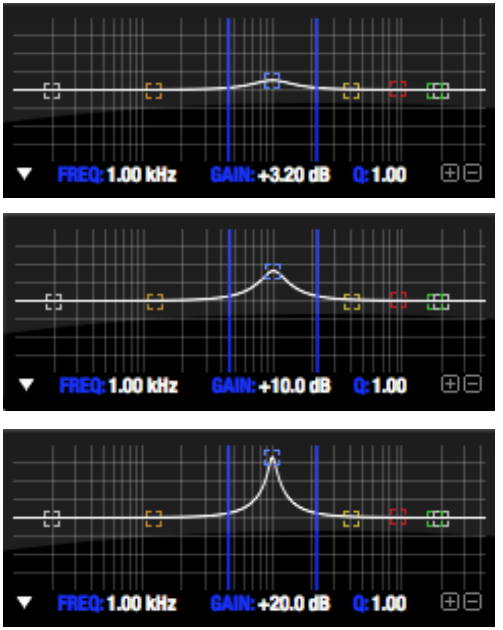


Figure 9-14: Type II EQ filter style.

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

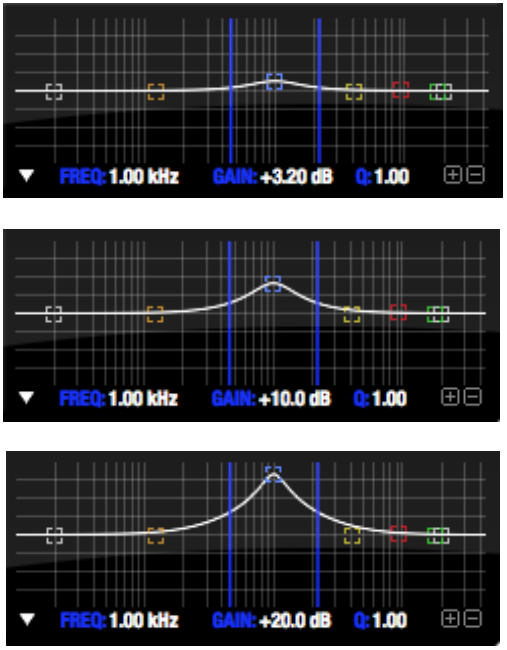


Figure 9-15: Type III EQ filter style.

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

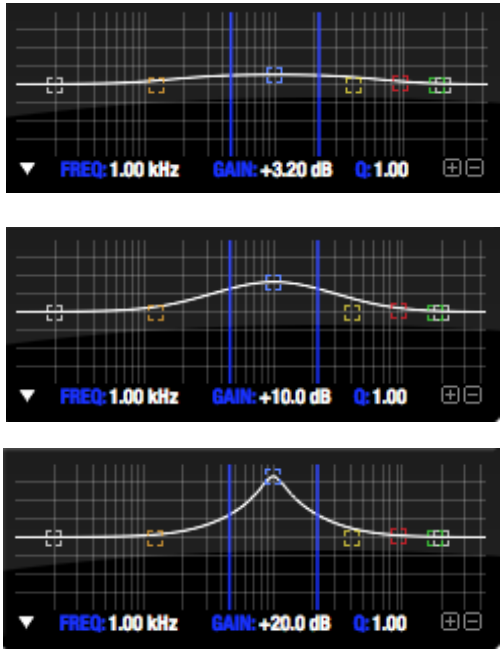


Figure 9-16: Type IV EQ filter style.

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 situations 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

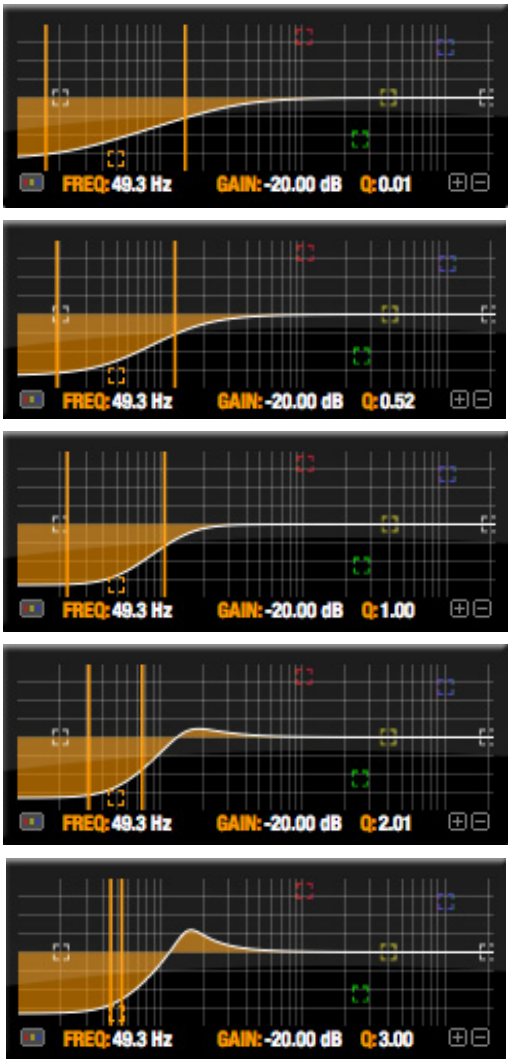


Figure 9-17: Shelf filter Q parameter overshoot.

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:

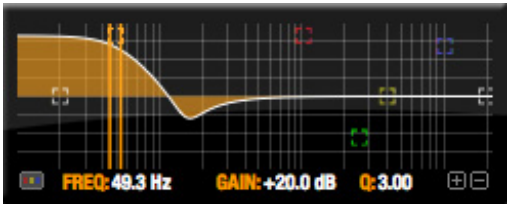


Figure 9-18: Overshoot when low shelf boost is applied.

Overshoot is also applied to high shelf boost and cut:

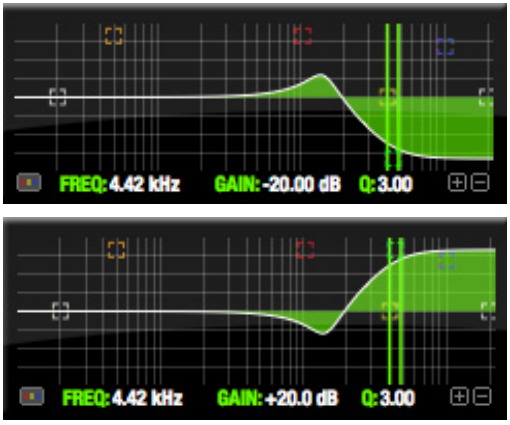


Figure 9-19: Overshoot when high shelf cut and boost is applied.

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 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.

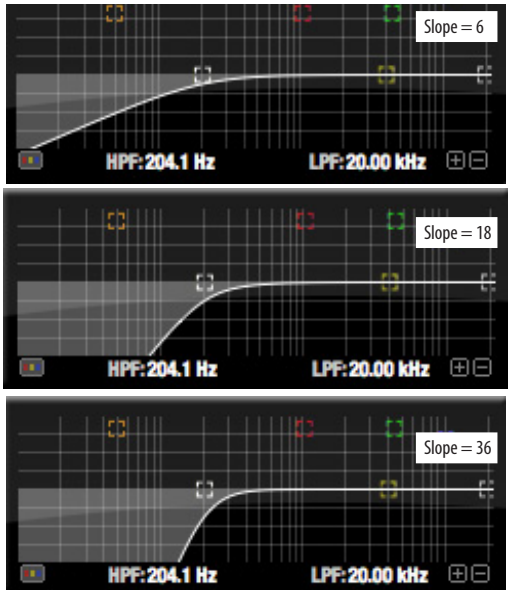


Figure 9-20: The low pass filter with three example slope settings.

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.

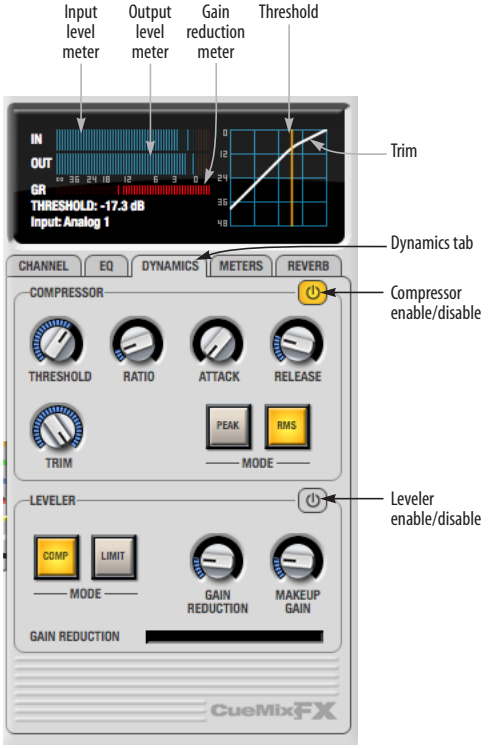


Figure 9-21: The Dynamics tab.

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. The dynamics processor is disabled when the 828x is operating at a 4x sample rate (176.4 or 192 kHz).

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 types 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 828x 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 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 828x. This tab gives you a "bird's-eye" view of all signal activity in the 828x; it is ideal for confirming your signal routing programming and for troubleshooting.

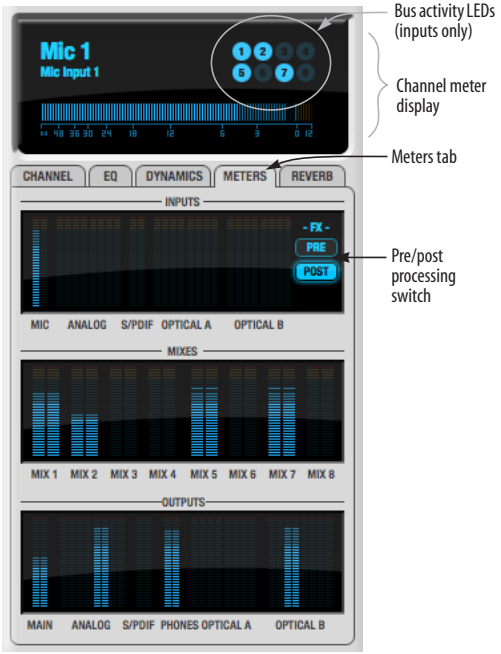


Figure 9-22: The Meters tab.

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 79.

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 828x's single, global reverb processor, which provides high-fidelity reverberation and graphic control over its parameters.

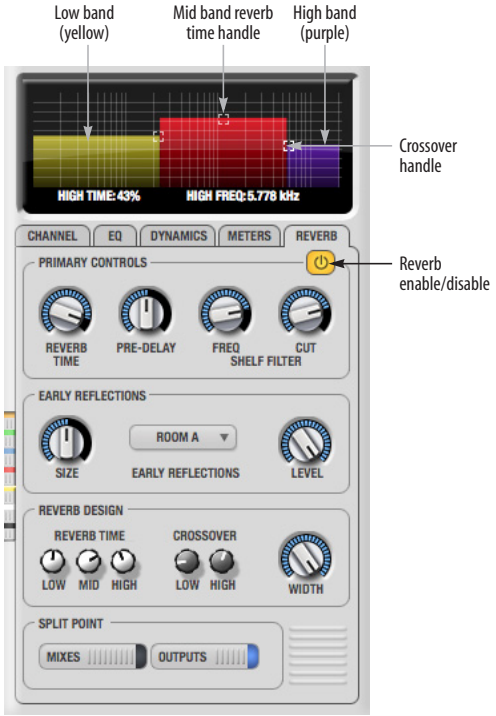


Figure 9-23: The Reverb tab.

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.

☛ The reverb processor is disabled when the 828x is operating at high sample rates (from 88.2 to 192 kHz).

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 74)
- Mix bus output (Figure 9-2 on page 72)
- Outputs (Figure 9-6 on page 77)

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.

Mixes

When the Split Point is set to *Mixes*, 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.

Outputs

When the Split Point is set to *Outputs*, 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 828x 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 MAIN VOL knob on the 828x front panel.

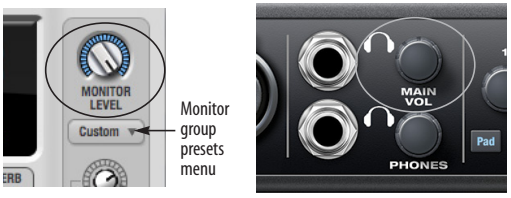


Figure 9-24: Monitor group volume control from CueMix FX and the front panel MAIN VOL knob.

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 77).

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 words *user def.* (user defined).

Monitor group meters

The monitor group meters (Figure 9-1 on page 71) 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 828x 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 828x output, as demonstrated below in Figure 9-25.

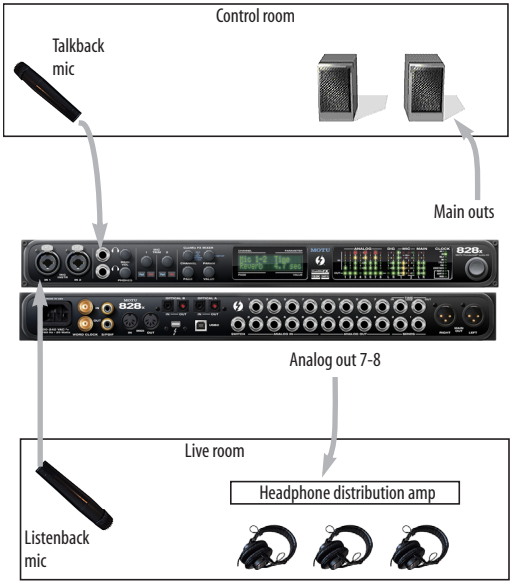


Figure 9-25: Typical hardware setup for Talkback and Listenback.

Talkback / Listenback Mic Input

To configure the talkback mic in CueMix FX, go to the Inputs tab (Figure 9-3 on page 74) and click the Focus button for the input that the talkback mic is connected to. Click the Channel tab (Figure 9-8 on page 78) 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 77) 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 Talkback 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. Option/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.
Command key	Applies your action to the stereo input pair, even when it is currently configured as mono.
Option key	Applies your action to all busses.
Shift-Option	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.
Option-click	Applies EQ enable button changes to all bands in that input or output.
Shift-Option-click	Applies EQ enable button changes to all bands and all inputs or outputs.

FILE MENU

Saving and loading hardware presets

The 828x 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 828x.

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 828x.

Hardware follows CueMix Stereo Settings

This File menu item applies to other MOTU interfaces products and has no effect on the 828x.

Show meter in dock icon

This CueMix FX File menu item applies to other MOTU interfaces and has no effect on the 828x.

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 Command-C).

2 Choose the destination mix and choose Paste from the Edit menu (or press Command-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.

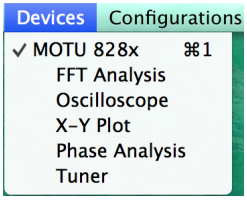


Figure 9-27: Opening the signal analysis windows.

Choosing channels for audio analysis

The audio analysis tools follow the currently focused audio input or output. (See "Channel focus and settings" on page 70.) 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 828x 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 80) or as a separate window (Figure 9-29 on page 96) opened from the Devices menu (Figure 9-27).

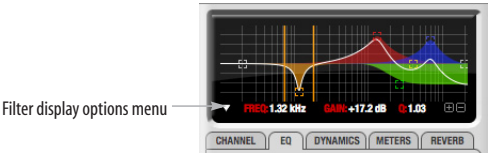


Figure 9-28: Filter Display options menu

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 beneath EQ filter points.
Show/Hide Full Window Analysis	Shows/hides the enlarged filter display in the CueMix FX window.

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:

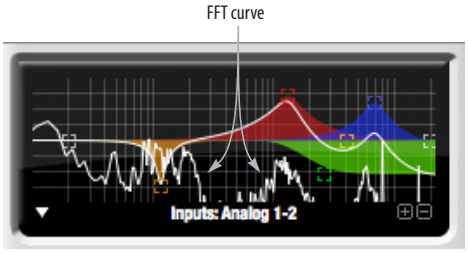


Figure 9-30: FFT display.

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:

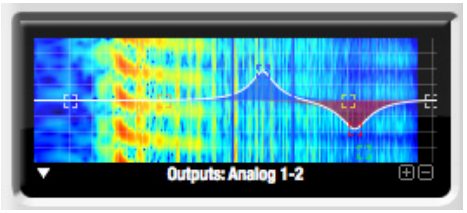


Figure 9-31: FFT display.

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 (Figure 9-27) to open a new window with the filter EQ display for detailed inspection and adjustment of the EQ filter, as shown Figure 9-29.

View controls

You can show and hide the FFT display, spectrogram or EQ band response curves as desired using the *View* controls (Figure 9-33).

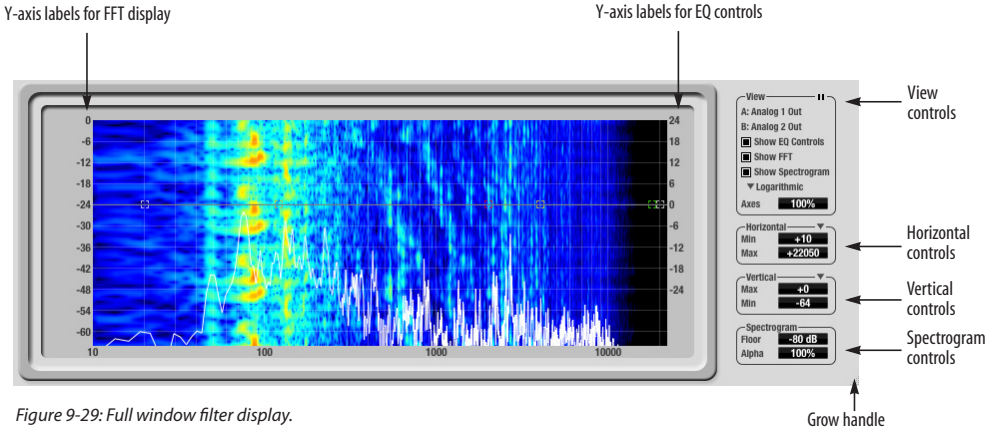


Figure 9-29: Full window filter display.

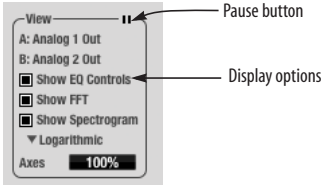


Figure 9-33: View controls

These settings are independent of the small graph display options (Figure 9-10 on page 80), so you have the flexibility to display different combinations in each graph.

☛ "Show EQ Controls" will be available only if the focused pair is 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.

Horizontal controls (frequency axis)

The *Horizontal* controls (Figure 9-34) 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 9-34).

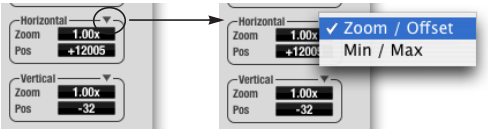


Figure 9-34: Horizontal control menu

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).

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 9-35).

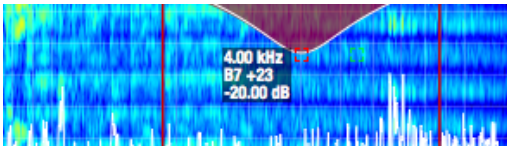


Figure 9-35: The Info Box.

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.

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 828x 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.

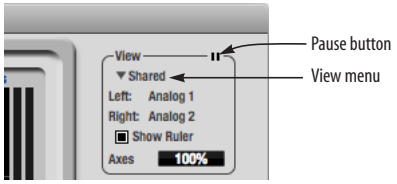


Figure 9-37: View controls

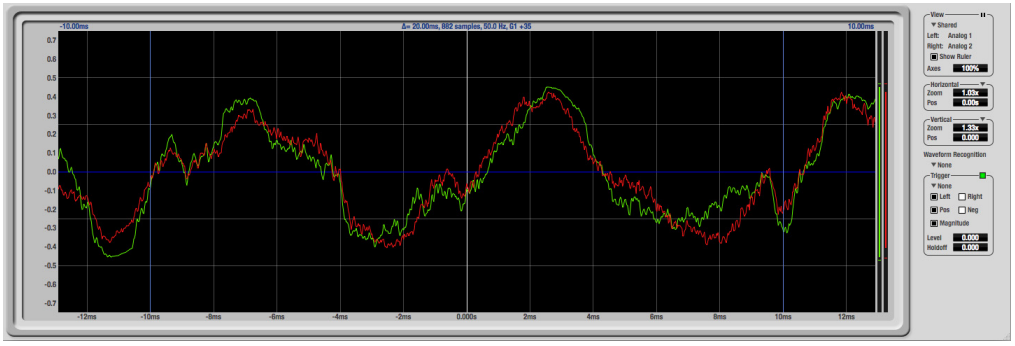


Figure 9-36: Oscilloscope

View menu

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

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

Display options

The *Axes* control (Figure 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 101).

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).

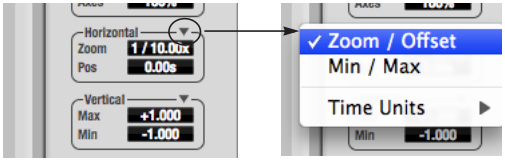


Figure 9-38: Horizontal control menu

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. *Pos* 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 *Pos* 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.

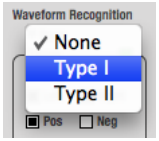


Figure 9-39: Waveform Recognition menu

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.

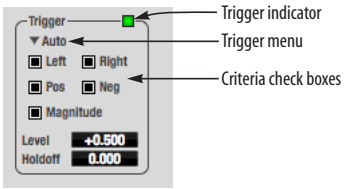


Figure 9-40: Trigger settings

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 100) 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 100) or by pressing the spacebar.

Trigger indicator

The Trigger indicator (Figure 9-40 on page 100) 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 *None*).
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 99).
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.

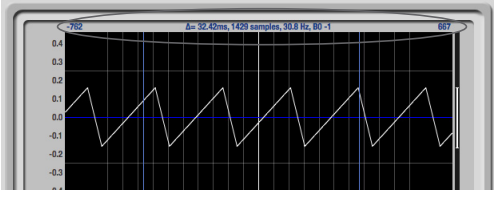


Figure 9-41: Measurement information

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

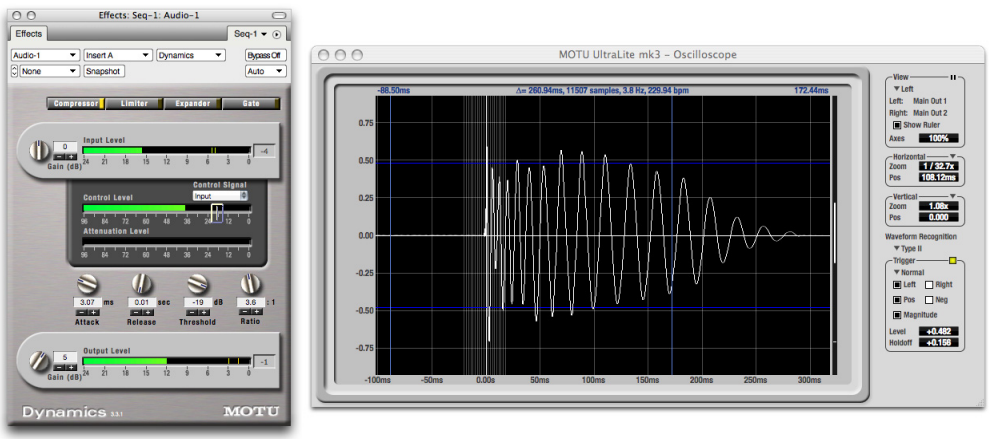


Figure 9-42: Viewing transients in the Oscilloscope

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 100), 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 tracked 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 828x interface has its own X-Y Plot window. Choose the *X-Y Plot* item from the Devices menu under the desired interface.

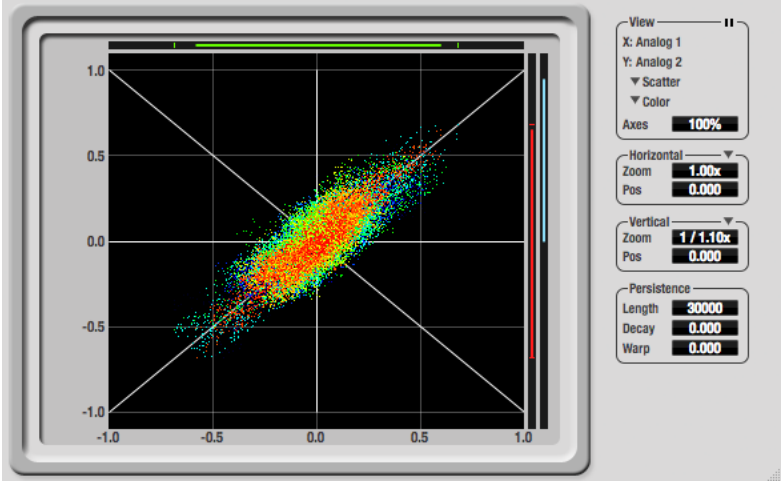


Figure 9-43: X-Y Plot

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).

View controls

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

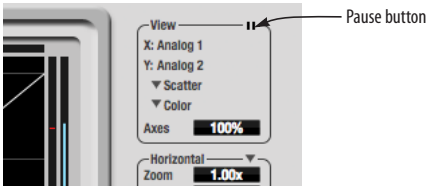


Figure 9-44: View controls

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.

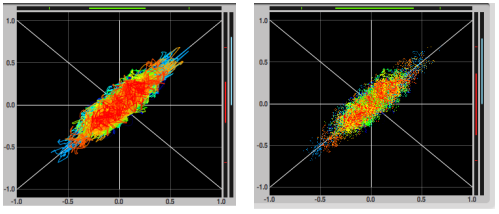


Figure 9-45: The same X-Y Plot displayed in Line versus Scatter mode.

☛ 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 106.

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.

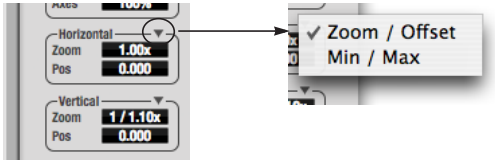


Figure 9-46: Setting the Horizontal or Vertical control modes.

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.

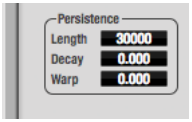


Figure 9-47: The Persistence controls.

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 106) 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.

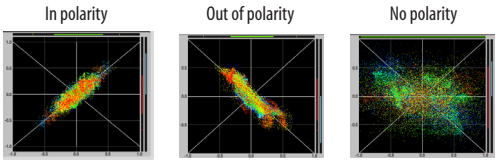


Figure 9-48: Checking polarity in a stereo signal with the X-Y Plot.

PHASE ANALYSIS

The *Phase Analysis* window (Figure 9-49 on page 107) graphs frequency versus phase difference versus amplitude of a stereo signal on either rectangular or polar coordinates.

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

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

Correlation Meter

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

Opening the Phase Analysis

Each MOTU audio interface has its own Phase Analysis window. Choose the *Phase Analysis* item from the Devices menu under the desired interface (Figure 9-27 on page 95).

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.

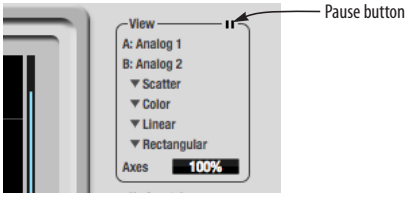


Figure 9-50: View controls

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.

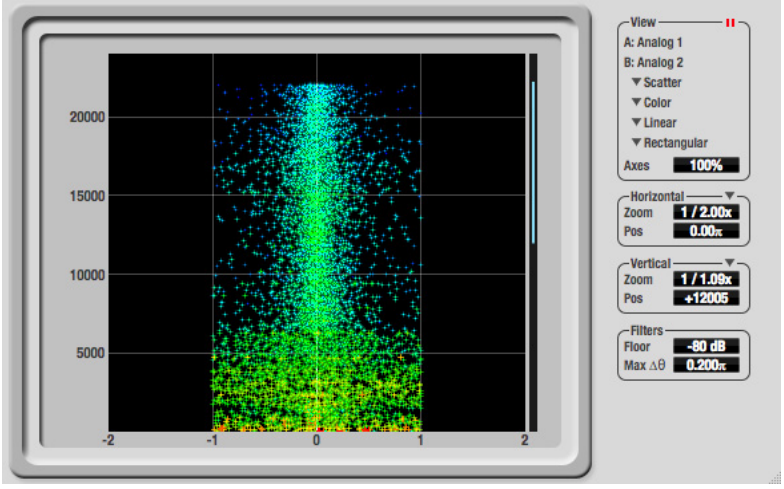


Figure 9-49: Phase Analysis

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.

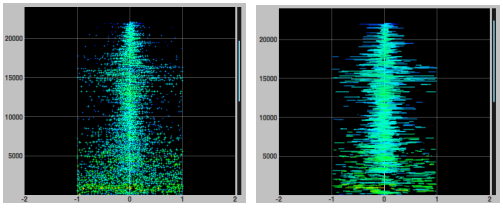


Figure 9-51: The same Phase Analysis displayed in Line versus Scatter mode.

☛ 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 109).

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.

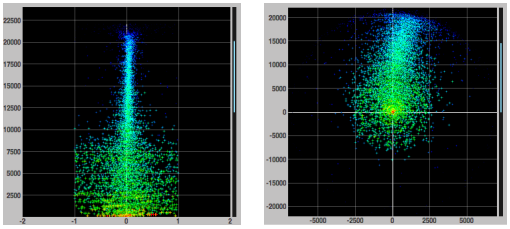


Figure 9-52: Rectangular versus Polar display (with a linear plot).

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:

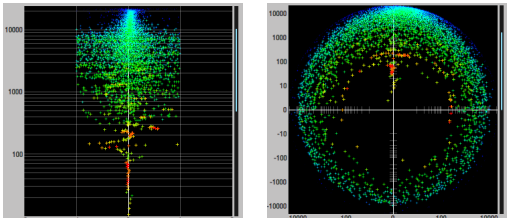


Figure 9-53: Rectangular versus Polar display with a logarithmic plot.

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.

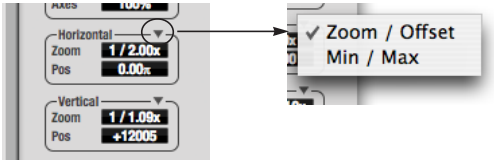


Figure 9-54: Setting the Horizontal or Vertical control modes.

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.

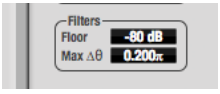


Figure 9-55: Filters

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 108) 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 110), 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 110.

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) 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).

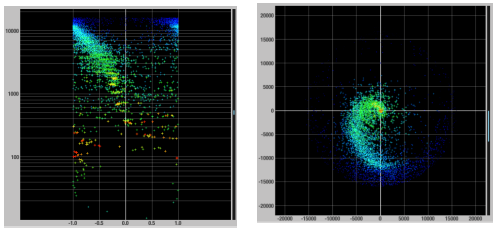


Figure 9-57: A stereo mix with phase issues.

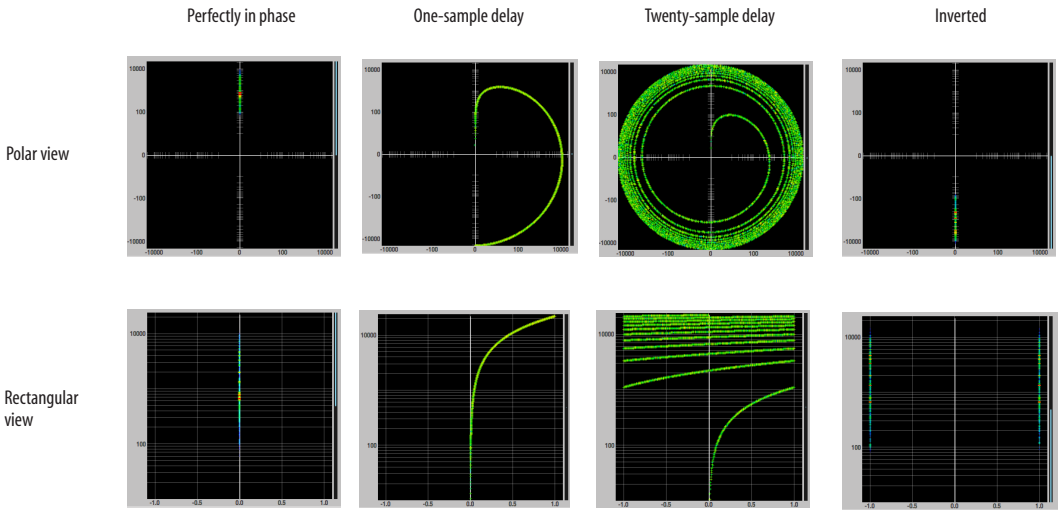
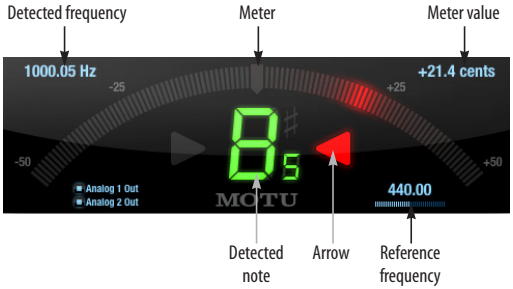


Figure 9-56: Two identical audio streams in the Phase Analysis.

TUNER

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



Opening the Tuner

Each MOTU audio interface has its own Tuner window. Choose the *Tuner* item from the Devices menu under the desired interface (Figure 9-27 on page 95).

Choosing a channel to tune

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).

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

Tuner controls

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

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

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

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

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

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

Tuning stereo signals

When tuning a stereo signal, the tuner analyzes the sum of the two channels. If the channels are not phase coherent, the tuner may not be able to measure the frequency of the signal. To tune only one channel of the channel pair, disable one of the channels as described in “Choosing a channel to tune” on page 111.

CONFIGURATIONS MENU

A configuration is just like a hardware preset (a "snapshot" of all settings in CueMix FX and therefore the 828x hardware itself), except that it can be created and managed using the CueMix FX software on your computer, completely independently of the 828x 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 Setup. 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.

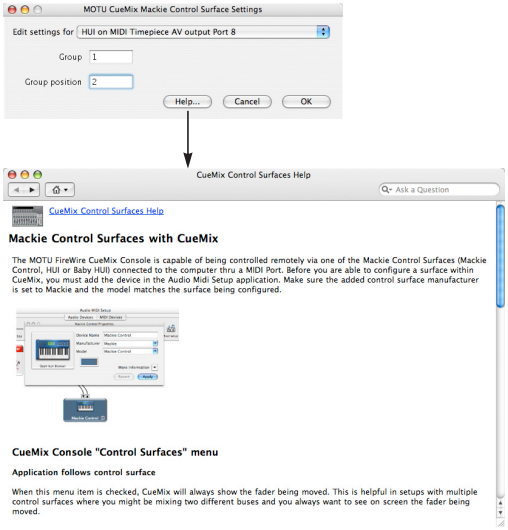


Figure 9-58: Refer to the extensive on-line help for details about configuring CueMix FX for operation with your control surface product.

Other HUI-compatible control surfaces

Any control surface that has the ability to emulate a HUI should be compatible with CueMix FX. Just add a Mackie HUI to Audio MIDI Setup and put the control surface hardware into HUI emulation mode. Consult the control surface manual for details about how put it into HUI emulation mode.

Other control surface hardware products

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

CHAPTER 10 MOTU SMPTE Setup

OVERVIEW

The 828x can resolve directly to SMPTE time code via any analog input, without a separate synchronizer. The 828x can also generate time code via its time code output. The 828x 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 AudioDesk and Digital Performer on Mac OS X. Other hosts, such as Pro Tools, can resolve to MIDI Time Code generated by the 828x.

MOTU SMPTE SETUP

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

CLOCK/ADDRESS

The *Clock/Address* menu (Figure 10-1) provides the same global *Clock Source* setting as in MOTU Audio Console ("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 828x 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 828x can auto-detect and switch to the incoming frame rate, except that

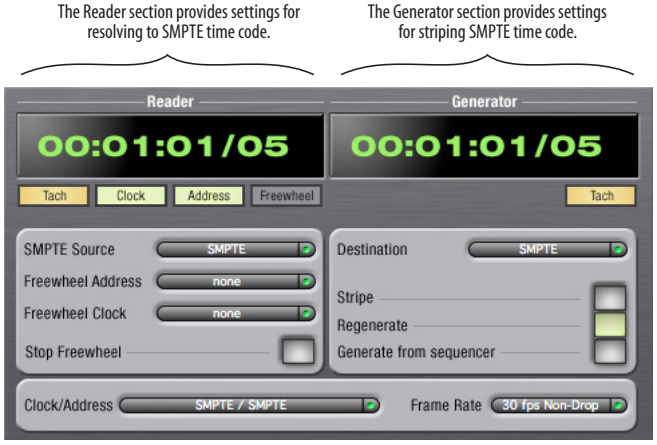


Figure 10-1: SMPTE Setup gives you access to your 828x's on-board SMPTE time code synchronization features.

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. The 828x driver updates the frame rate setting in Digital Performer and AudioDesk for you.

READER SECTION

The Reader section (on the left-hand side of the window in Figure 10-1) provides settings for synchronizing the 828x 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 828x has successfully achieved lockup to SMPTE time code and SMPTE frame locations are being read.

Clock

The Clock light glows continuously when the 828x 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 828x has successfully achieved lockup to SMPTE time code.

Freewheel

The Freewheel light illuminates when the 828x 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 828x "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 828x 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 828x to freewheel before it gives up and stops altogether.

The 828x 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 828x 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 a glitch or drop-out in the incoming SMPTE time code for some reason. The 828x can freewheel past the drop-out and then resume lockup again as soon as it receives a stable, readable clock signal.

The 828x 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 828x 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 828x. The level knob disappears when the Destination is set to *None*.

Tach light

The Tach light blinks once per second when the 828x 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.

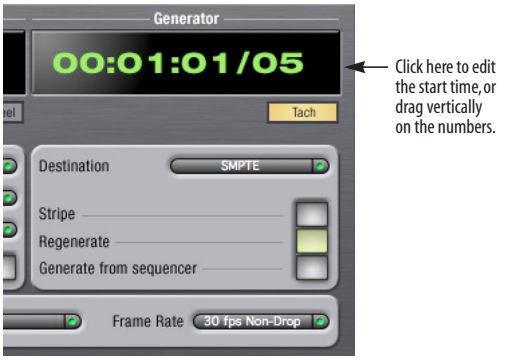


Figure 10-2: Setting the time code start time.

Regenerate

This option, when enabled, causes the generator to generate time code whenever the 828x is receiving SMPTE time code.

Generate from sequencer

This option, when enabled, causes the generator to generate time code whenever you are running AudioDesk or Digital Performer. Time code begins at the time specified by the AudioDesk or Digital Performer main transport.

SETTING UP FOR SMPTE TIME CODE SYNC

To set up direct SMPTE time code synchronization, see "Syncing to SMPTE timecode" on page 33.

RESOLVING DP OR AUDIODESK TO TIME CODE

To resolve Digital Performer or AudioDesk directly to time code with no additional sync devices, use the setup shown in "Syncing to SMPTE timecode" on page 33. Choose *Receive Sync* from the Setup menu and choose the *Sample accurate* option. Make sure that the *Slave to External Sync* command in the Studio menu is checked. Make sure the *Clock Source* setting in the MOTU Audio Setup window is set to *SMPTE*. Also, make sure that you've connected an LTC input signal to the 828x time code input, and that you've specified that input in *SMPTE Source* menu in SMPTE Setup.

RESOLVING PRO TOOLS TO TIME CODE

To resolve your Pro Tools system directly to SMPTE time code with no additional synchronization devices, use the setup shown in "Syncing to SMPTE timecode" on page 33.

1 Connect an LTC signal to the 828x time code input (or other analog input).

2 In MOTU SMPTE Setup, go to the SMPTE Source menu and choose the *SMPTE* input (or other analog input that is receiving the time code).

3 In MOTU SMPTE Setup, go to the Clock Source menu and choose *SMPTE/SMPTE* as the clock source.

4 Make the other settings in MOTU SMPTE Setup as desired. Refer to their sections in this chapter for more information.

5 In Pro Tools, choose *Peripherals* from the Setup menu and click the Synchronization tab.

6 Choose the 828x Sync Port from the MTC Reader Port menu.

RESOLVING OTHER HOSTS TO TIME CODE

The 828x has the ability to convert incoming SMPTE time code (LTC) to MIDI Time Code and send it to your host audio software, which can resolve to it. To do so, use the setup shown in "Syncing to SMPTE timecode" on page 33. Here is the basic procedure:

1 Connect an LTC signal to the 828x time code input (or other analog input).

2 In MOTU SMPTE Setup, go to the SMPTE Source menu and choose the *SMPTE* input (or other analog input that is receiving the time code).

3 In MOTU SMPTE Setup, go to the Clock Source menu and choose *SMPTE/SMPTE* as the clock source.

4 Make the other settings in SMPTE Setup as desired. Refer to their sections in this chapter for more information.

5 In your host audio software, make the necessary settings for resolving it to MIDI Time Code. Refer to your host audio software documentation for further information. The 828x driver provides a separate MIDI Sync Port for the time code, so when you specify the MIDI port to follow, be sure to choose the Sync Port.

Part 3
Appendices

APPENDIX A Troubleshooting

Slaving directly to time code in AudioDesk or Digital Performer
To slave Digital Performer or AudioDesk directly to time code, be sure to go to the Receive Sync dialog in Digital Performer or AudioDesk and switch from "MTC" to "Sample-accurate."

Can't authenticate AudioDesk
When authenticating AudioDesk, the OK button does not become active until you have entered in your name and a valid keycode. Your name must contain at least three characters, and you must enter the keycode exactly as it appears (on the jacket of your AudioDesk installer disc). If you continue to have difficulties, try repairing Mac OS X disk permissions using *Disk Utility*.

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 32 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 Clock Source to *Internal* and try recording just using the analog inputs and outputs of the 828x. 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.

Connecting or powering gear during operation
It is not recommended that you connect/disconnect, or power on/off devices connected to the 828x while recording or playing back audio. Doing so may cause a brief glitch in the audio.

828x inputs and outputs are not available in host audio software
Make sure that the inputs and outputs are enabled. See "Working with 828x inputs and outputs" on page 59.

No optical inputs or outputs are available in host audio application
Check to make sure you have the desired optical inputs and/or outputs enabled in the MOTU Audio Setup.

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 828x's hardware-based CueMix FX monitoring feature. Please see chapter 8, "Reducing Monitoring Latency" (page 63).

Controlling monitoring latency
See chapter 8, "Reducing Monitoring Latency" (page 63).

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 828x 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 828x system. This is printed on a sticker placed on the bottom of the 828x 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 828x 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 828x Development Team, MOTU Inc., 1280 Massachusetts Avenue, Cambridge, MA 02138.

APPENDIX B Audio I/O reference

OVERVIEW

The MOTU Audio drivers supply text string labels for the 828x'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.

Inputs at 1x sample rates

Inputs are always listed in the same order as follows, when operating the 828x at 1x sample rates (44.1 or 48 kHz):

Input	Channels	List position	Comment
Mic-Guitar	2	1-2	-
Analog	8	3-10	-
Reverb return	2	11-12	See "Reverb return" on page 56.
Stereo return	2	13-14	See "Return Assign" on page 56.
SPDIF	2	15-16	-
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 828x at a 2x sample rate (88.2 or 96 kHz), inputs are listed as follows:

Input 88.2 / 96 kHz	Channels	List position	Comment
Mic-Guitar	2	1-2	-
Analog	8	3-10	-
Reverb return	not available	not available	not available
Stereo return	2	11-12	See "Return Assign" on page 56.
SPDIF	2	13-14	-
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 828x at a 4x sample rate (176.4 or 192 kHz), inputs are listed as follows:

Input 176.4 /192 kHz	Channels	List position	Comment
Mic-Guitar	2	1-2	-
Analog	8	3-10	-
Reverb return	not available	not available	not available
Stereo return	2	11-12	See "Return Assign" on page 56.
SPDIF	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 828x at 1x sample rates (44.1 or 48 kHz):

Output 44.1 / 48 kHz	Channels	List position	Comment
Analog	8	1-8	-
Main outs	2	9-10	If the main outs are assigned to mirror another output pair (such as the analog 1-2), they won't be listed separately.
Phones	2	11-12	If the phones are assigned to mirror another output pair (such as the main outs), they won't be listed separately.
SPDIF	2	13-14	If the phones are mirroring, then subtract 2. If the main outs are mirroring, subtract another 2.
Optical A	8 ADAT 2 TOSLink	15-22 15-16	-
Optical B	8 ADAT 2 TOSLink	23-30 17-18	These channel number ranges assume that optical A is operating with the same format.

Outputs at 2x sample rates

When operating the 828x at a 2x sample rate (88.2 or 96 kHz), outputs are listed as follows:

Output 88.2/ 96 kHz	Channels	List position	Comment
Analog	8	1-8	-
Main outs	2	9-10	If the main outs are assigned to mirror another output pair (such as the analog 1-2), they won't be listed separately.
Phones	Mirror only	--	-
SPDIF	2	11-12	If the main outs are mirroring, then subtract 2.
Optical A	4 ADAT 2 TOSLink	13-16 13-14	-
Optical B	4 ADAT 2 TOSLink	17-20 15-16	These channel number ranges assume that optical A is operating with the same format.

Outputs at 4x sample rates

When operating the 828x at a 4x sample rate (176.4 or 192 kHz), outputs are listed as follows:

Output 176.4/ 192 kHz	Channels	List position	Comment
Analog	8	1-8	-
Main outs	2	9-10	-
Phones	Mirror only	--	-
SPDIF	Not available	--	-
Optical A	Not available	--	-
Optical B	Not available	--	-

A

B

C

D

E

F

V

W

X