16A™

User Guide

MOTU®

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SAFETY PRECAUTIONS AND ELECTRICAL REQUIREMENTS FOR THE 16A ("PRODUCT")

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 VOLTAGES: 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: THIS EQUIPMENT IS NOT SUITABLE FOR USE IN LOCATIONS WHERE CHILDREN ARE LIKELY TO BE PRESENT.

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 PRODUCT COULD CAUSE AN ELECTRICAL SHOCK.

The MOTU product 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 product 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. In the EU, use the supplied EU power cord and be sure that the power outlet is properly grounded (Figure C).

Figure A

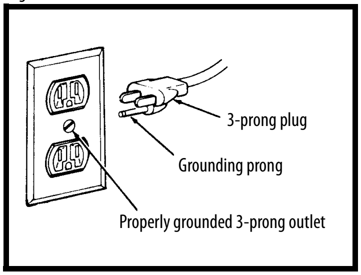


Figure B

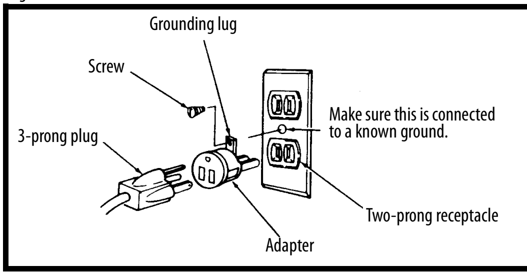


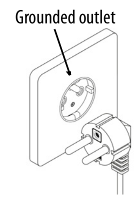
Figure C

Danmark: Apparatets stikprop skal tilsluttes en stikkontakt med jord som giver forbindelse til stikproppens jord.

Suomi: Laite on liitettävä suojakoskettimilla varustettuun pistorasiaan.

Norge: Apparatet må tilkoples jordet stikkontakt.

Sverige: Apparaten skall anslutas till jordat uttag.



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 product plug.

IMPORTANT SAFEGUARDS

The device with CLASS I construction must be connected to the mains socket with a protective earthing connection.

1. Read these instructions. All the safety and operating instructions should be read before operating the product.
2. Keep these instructions. These safety instructions and the product owner's manual should be retained for future reference.
3. Heed all warnings. All warnings on the product 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 product near water.
6. Cleaning - Unplug the product 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 product 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 product 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 button - Install the product so that the power button can be accessed and operated at all times.
13. Disconnect - The main plug is considered to be the disconnect device for the product 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 product. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
16. Surge protection - Unplug the product 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 product 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 product, the product 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 product 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 product, ask the service technician to perform safety checks to determine that the product is in safe operating conditions.

ENVIRONMENT, HEAT AND VENTILATION

Operating Temperature: 10°C to 40°C (50°F to 104°). The product should be situated away from heat sources or other equipment that produces heat. When installing the product in a rack or any other location, be sure there is adequate space around the product to ensure proper ventilation. Improper ventilation will cause overheating and can damage the unit.

TO REDUCE THE RISK OF ELECTRICAL SHOCK OR FIRE

Do not handle the power cord with wet hands. Do not pull on the power cord when disconnecting it from an AC wall outlet. Grasp it by the plug. Do not expose this apparatus to rain or moisture. Do not place objects containing liquids on it.

AC INPUT

100 - 240VAC ~ • 50 / 60Hz • 1.0A max

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UPDATE POLICY

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

COPYRIGHT NOTICE

Limited Warranty on Hardware

Mark of the Unicorn, Inc. ("MOTU") warrants this equipment against defects in materials and workmanship under normal use for a period of TWO (2) YEARS from the date of original retail purchase. The Warranty Term begins on the date of purchase from an authorized MOTU reseller and applies solely to the original retail purchaser, who must activate the warranty by creating a user account at motu.com to register the product within 90 days of purchase. This warranty applies only to hardware products; MOTU software is licensed and warranted pursuant to separate written statements.

If you discover a defect, first contact MOTU technical support by phone, email or web (motu.com/support) to verify the warranty on your MOTU equipment and obtain a Return Merchandise Authorization (RMA). No service will be performed on any product returned without prior authorization. MOTU will, at its option, repair or replace the product at no charge to you, provided you return it during the warranty period as instructed by MOTU, with transportation charges prepaid. If you purchased your equipment in any country other than the US or Canada, you will be instructed to return the equipment to an authorized MOTU distributor or representative in the country of purchase. You must use the product's original packing material for the shipment, and insure the shipment for the value of the product. Please include your name, address, phone number, email address, a description of the problem, and the original, dated bill of sale with the returned unit; do NOT include additional accessories such as cables, power supplies, manuals, etc. Please clearly print the Return Merchandise Authorization Number on the outside of the box below the shipping address. Repaired or replaced equipment will be returned to you via UPS Ground prepaid. (Expedited shipping methods such as UPS next day, 2-day, and 3-day services are available for an additional cost.) Repaired equipment will be warranted for a period equal to the remainder of the original Limited Warranty or for 90 days, whichever is longer.

WARRANTY EXCLUSIONS: This warranty does not apply if the equipment has been damaged by accident, abuse, misuse, or misapplication; has been modified without the written permission of MOTU; or if the product serial number has been removed or defaced. The following examples, without limitation, are NOT covered by this hardware warranty:

- Equipment purchased through any reseller not directly authorized by MOTU or its authorized international distributors.
- "Used" equipment purchased from a third party.
- Equipment purchased in another country.
- Normal cosmetic and mechanical wear of the equipment.
- Equipment damaged by improper installation or connections.
- Equipment damaged in transit to/from MOTU for warranty repair.
- Physically damaged equipment, including but not limited to water damage, cracks or dents, missing or bent parts, burns or other damage caused by faulty or failed electric power

ALL IMPLIED WARRANTIES, INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE, ARE LIMITED IN DURATION TO TWO (2) YEARS FROM THE DATE OF THE ORIGINAL RETAIL PURCHASE OF THIS PRODUCT. THE WARRANTY AND REMEDIES SET FORTH ABOVE ARE EXCLUSIVE AND IN LIEU OF ALL OTHERS, ORAL OR WRITTEN, EXPRESS OR IMPLIED. No MOTU dealer, agent, or employee is authorized to make any modification, extension, or addition to this warranty. MOTU IS NOT RESPONSIBLE FOR SPECIAL, INCIDENTAL, OR CONSEQUENTIAL DAMAGES RESULTING FROM ANY BREACH OF WARRANTY, OR UNDER ANY LEGAL THEORY, INCLUDING LOST PROFITS, DOWNTIME, GOODWILL, DAMAGE OR REPLACEMENT OF EQUIPMENT AND PROPERTY AND COST OF RECOVERING REPROGRAMMING, OR REPRODUCING ANY PROGRAM OR DATA STORED IN OR USED WITH MOTU PRODUCTS.

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

Version 1.00

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

• Relocate or reorient the receiving antenna

• Increase the separation between the equipment and the receiver

• Plug the equipment into an outlet on a circuit different from that to which the receiver is connected

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

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

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

Contents

Part 1
Getting Started

Quick Start Guide

Thank you for purchasing a 16A! Follow these easy steps to get started quickly.

MAC USERS START HERE

1 Visit *motu.com/16A-start* to download the *MOTU Pro Audio V2.dmg* virtual disk image.

2 Copy the CueMix Pro app to your Applications folder. If you are running macOS 13 or later, launch CueMix Pro and click *Install* in the *Discovery* tab. Visit *System Settings > Privacy & Security* to allow the driver to intall.

3 Connect the included power cord to the 16A.

4 Connect the 16A to your Mac using the included USB-C cable. If your Mac requires it, use a USB-IF certified USB-C-to-A cable. Plug adapters are not recommended. See page 18.

5 Choose *Apple menu > System Settings (or System Preferences)* and click *Sound* to choose the *16A* as the input and output device.

6 Proceed to “For all users”.

WINDOWS USERS START HERE

1 BEFORE you connect the 16A to your computer, visit *motu.com/16A-start* to download and run *MOTU Pro Audio V2* installer.

2 Connect the included power cord to the 16A.

3 Connect the 16A to your computer using the included USB-C cable. If your computer requires it, use a USB-IF certified USB-C-to-A cable. Plug adapters are not recommended. See page 18.

4 Go to the Windows Sound Control Panel and choose *16A* as the default recording and playback device.

5 Proceed to “For all users”.

FOR ALL USERS

6 As shown on the next page, connect speakers to the 16A *Line Out 1-2*, or connect a pair of headphones to the headphone output on the front panel, so you can hear your computer’s audio output.

7 You are now ready to start using your 16A interface.

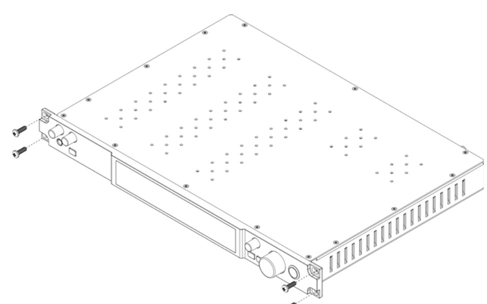
8 Visit *motu.com/16A-start* to register your 16A, download the included software and watch brief how-to videos, including:

- How to connect other gear to the 16A.
- How to use the 16A with your recording software.
- How to get the most out of the 16A.

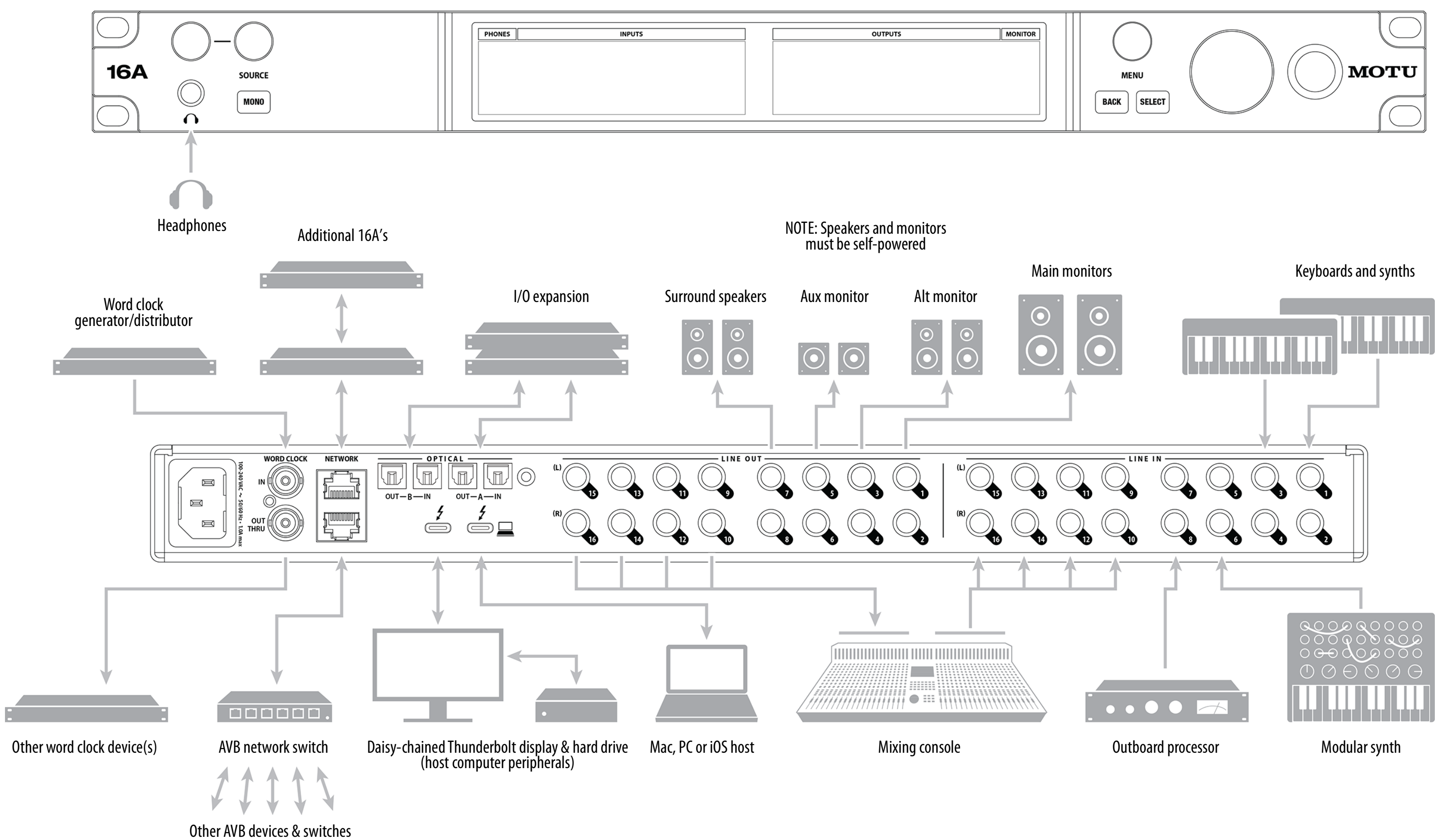
☛ Register your 16A to gain access to all the software, virtual instruments, loops and sounds that are included with your 16A purchase. Registered users also qualify for technical support and information about software updates, so please register today!

RACK-MOUNT INSTALLATION

The 16A can be mounted in a standard 19-inch (48.26 cm) equipment rack using four standard 10-32 5/8th-inch or M6 1mm pan head rack screws. It is one rack unit (1U) high (1.75 inches, 44.45 mm) with a depth of 12.25 inches (31 cm).



A Typical 16A Setup



16A Front Panel

1. Connect HEADPHONES here and use the volume knob to control the volume level independently from other outputs. As you turn the knob, the LCD provides visual feedback with a volume meter.

2. Choose the SOURCE for the headphone signal here. The PHONES section (3) displays the currently selected source.

3. The PHONES section provides metering, the current volume setting, and the currently selected signal source for the headphones. Use volume (1) and source select (2) to control the level and signal source.

4. The INPUTS section of the color LCD displays metering for all 16A analog and digital (optical) inputs. Use the *Meter View* MENU setting (7) to choose among several different meter arrangements.

5. The OUTPUTS section of the color LCD displays metering for all 16A analog and digital (optical) outputs. Use the *Meter View* MENU setting (7) to choose among several different meter arrangements.

6. The MONITOR section shows the current level for the Monitor Group, as controlled by the main volume knob (8).

7. Turn the MENU knob (7) to enter the LCD menu and scroll through menu options. Push SELECT (10) to go into the sub-menus, if applicable. To choose the current setting, push SELECT. Use the BACK button (11) to return to the previous menu level. Push BACK repeatedly to return to the main screen.

8. The large monitor knob controls the volume of Line Outs 1-2. This knob also supports surround monitoring. See “The Monitor Group” on page 28.

9. This is the POWER switch for the unit.

10. Push SELECT (or turn the MENU knob) to enter the menus. Push SELECT to select or confirm the currently selected menu setting. Push BACK to go back one level.

11. Push BACK to go back one level in the menu. When the display is showing meters, push BACK to clear the peak indicators and clip indicators, if any.

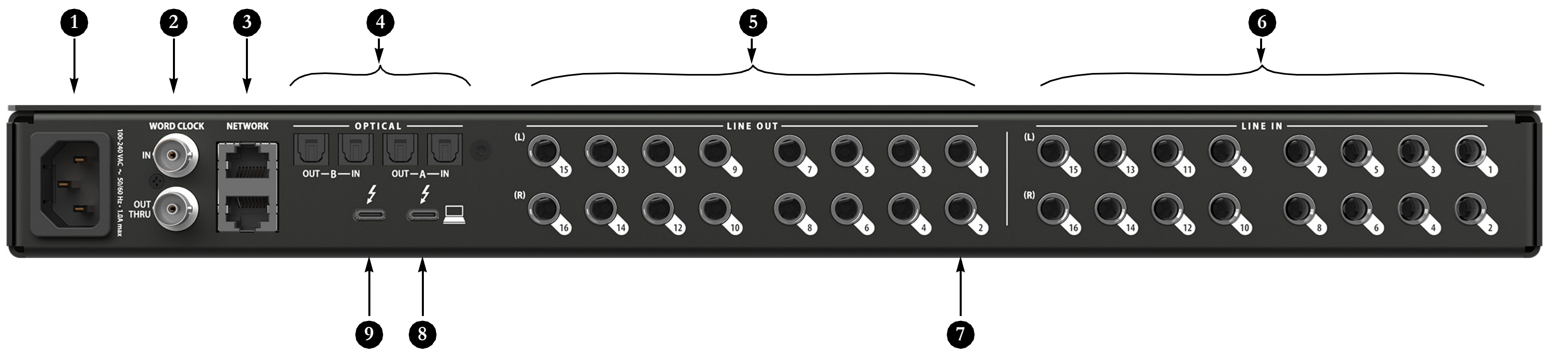
12. The *status section* displays the 16A’s current operating sample rate and clock source. The NETWORK icons show activity to and from the network. The HOST icon indicates the status of the host computer (or iOS device) connection. Blue indicates a successful connection to the host. The 16A negotiates the highest performance connection to the host. If a Thunderbolt or USB4 connection is established, the HOST icon displays *TB*. If a USB 2 or 3 connection is established, a *2* or *3* is displayed, depending on the USB format supported by the host computer. See “HOST Status” on page 27.

13. The headphone source signal is indicated here, as chosen with the SOURCE select knob (2).

14. When MONO is disabled, the headphone source signal is stereo, and the SOURCE select knob (2) lets you choose stereo pairs (e.g. Line Outs 5-6). When MONO is enabled, the headphone source signal is mono (split to left and right), and the SOURCE select knob (2) lets you choose individual mono channels (e.g. Line In 4).

15. Connect headphones here. Signal output is driven by ESS conversion, with plenty of gain.

16A Rear Panel



1. The 16A is equipped with a universal international power supply (100-240 VAC, 50/60 Hz, 1.0A max) with an standard IEC connector.

2. These are standard BNC WORD CLOCK jacks. Use them for a variety of applications, such as digital I/O with devices that cannot resolve to the clock supplied by their digital I/O connection with the 16A. See “Syncing word clock devices” on page 23.

3. These two NETWORK ports provide industry standard IEEE 802.1 (AVB) network connectivity to other network devices. Examples include:

   - Another 16A or any other MOTU AVB-equipped audio interface, such as the 1248, 8M, 16A (original), 24Ai, 24Ao, 112D, Monitor 8, etc.
   - A standard Ethernet hub or Wi-Fi router (for internet connection).
   - A standard AVB Ethernet switch for high-speed, low latency, high-capacity audio connectivity to an AVB audio network.

   For details see chapter 9, “Networking” (page 58).

4. These two banks of ADAT optical “lightpipe” connectors each provide 8 channels of 24-bit ADAT optical digital I/O at 1x sample rates (44.1 or 48 kHz) and 4 channels at 2x sample rates (88.2 or 96 kHz). They are disabled at higher sample rates. Bank A can alternately operate as stereo TOSLink (optical S/PDIF) connectors. See “Optical I/O” on page 21 and “Syncing optical devices” on page 23.

   Note: you can choose independent formats for Bank A IN and OUT. For example, you could choose ADAT for the optical A IN (for eight channels of input from your digital mixer, for example) and stereo TOSLink for the optical A OUT (for a set of auxiliary speakers, for example).

5. The sixteen LINE OUT jacks are balanced, DC-coupled quarter-inch connectors that can also accept a TRS plug with the ring disconnected for unbalanced operation (see “TRS quarter-inch line inputs” on page 21 for more on this). They provide analog output for studio monitors, surround monitoring, sub-mixes or any other desired destination. The output trim can be adjusted from the Outputs Tab in the CueMix Pro app. For surround monitoring, connect your surround speakers to outputs and see “The Monitor Group” on page 28.

6. These sixteen LINE IN jacks are balanced (TRS) quarter-inch connectors that can also accept an unbalanced TS plug. Use with line level analog signals up to +21 dBu, including synthesizers, drum machines, effects processors, etc. These inputs are also equipped with the 16A Precision Digital Gain™ feature: 0 to +20 dB of gain adjustable in 1 dB increments from the included CueMix Pro app (Inputs tab).

7. These two line out jacks serve as the 16A main output pair for primary (powered) studio monitors, PA speakers or any other desired destination. You can control their volume from the front panel MONITOR knob (#8 on page 9).

   To hear audio playback from your host audio software on this pair of outputs, assign audio tracks (and master fader) to Line Out 1-2. You can also use the CueMix Pro app to route live 16A inputs here as well.

8. Use this (upstream) HOST port to connect the 16A to a computer or iOS device using the included USB-C cable. The 16A negotiates the best-possible connection to the host (Thunderbolt, USB4, USB3 or USB2). For details, see chapter 4, “Hardware Installation” (page 18).

9. Use this (downstream) DEVICE port to connect computer peripherals such as hard drives, monitors, additional 16A’s, a USB hub or Thunderbolt dock, etc. Essentially, you can connect any device here that you could connect directly to your computer. If your computer supports Thunderbolt, you can daisy-chain multiple Thunderbolt devices to this port.

CHAPTER 1 About the 16A

The 16A is a 32 x 34 Thunderbolt/USB audio interface with mixing, DSP effects, AVB networking, and very high quality A/D/A conversion at sample rates up to 192 kHz.

Hardware-driven DSP delivers a powerful 64 channel monitor mixer with 16 stereo buses, 32-bit floating point effects processing, double-precision 4-band EQ, compression, gating, HPF, and reverb.

A comprehensive patch bay provides flexible routing between the mixer, local I/O and the AVB network.

The 16A can operate as an audio interface for a computer or iOS device, as a stand-alone digital mixer, as a gateway to an expanded studio system, as a component of an extended AVB audio network, or as a capable hybrid device performing all of these roles simultaneously.

The 16A is designed to be a central component of a modern, high performance recording studio or live mixing platform. The following sections provide a brief overview of its main features and characteristics.

Comprehensive I/O

The 16A provides 66 channels of simultaneous input and output.

Connection	Input	Output
Quarter-inch analog on bal/unbal TRS	16	16
Headphone output	-	Stereo
ADAT optical digital (at 44.1 or 48 kHz)†	16	16
Total	**32**	**34**

† The 16A optical connectors support the industry-standard ADAT and TOSLink optical I/O formats, which provide varying channel counts. TOSLink is available on Bank A, and ADAT is available on both banks (A and B). See “Optical I/O” on page 21 for details about optical bank operation.

Universal connectivity

The 16A connects to a host computer via Thunderbolt or USB, depending on the host. The 16A negotiates the best-possible connection to the host (Thunderbolt, USB4, USB3 or USB2). It is USB audio class-compliant, which means that it is iOS compatible and does not require driver installation for USB connection to macOS or iOS hosts. Industry standard audio drivers for both Thunderbolt and USB operation provide universal compatibility with any audio software.

Up to 256 channels of audio I/O

The 16A lets you stream up to 128 audio channels in and out, simultaneously, through its Thunderbolt connection to a host computer. Sources and destinations can include inputs and outputs on the device, inputs and outputs on other interfaces connected via AVB networking, and even audio software apps running on other computers connected to other devices on the network.

State-of-the-art A/D and D/A conversion

The analog section of each interface employs state-of- the-art 32-bit DACs and ADCs, which deliver analog recording and playback with remarkably high dynamic range at sample rates from 44.1 to 192 kHz.

Flexible analog I/O with Precision Digital Gain & Trim™
All quarter-inch analog inputs can accept either a balanced or unbalanced plug. The sixteen line inputs are equipped with 0 to +20 dB of digital gain, adjustable in 1 dB increments.

Equipped with renowned ESS Sabre32™ DAC technology, all analog outputs offer trim (cut), also adjustable in 1 dB increments. Outputs can be trimmed down to -99 dB.

All quarter-inch outputs are DC-coupled, so they can be used for control voltage (CV) output.

On-board DSP with mixing and processing
The 16A is equipped with a DSP engine that drives a 64-input monitor mixer with 32 buses. The mixer offers familiar operation modeled after large format mixing consoles. Effects include 4-band parametric EQ, compression, gating, high-pass filter, and reverb. The included CueMix Pro app provides easy and intuitive on-screen control of everything from your computer or iOS device.

32-bit processing
The DSP engine delivers 32-bit floating point processing for virtually infinite headroom, with 64-bit double-precision processing for the EQ filters for the utmost in sonic quality.

Software control
Control the 16A on-board mixing and device settings from the CueMix Pro app software running on a laptop or iOS device.

Stand-alone mixing
Connect an iOS device to the 16A for complete control of all settings on the road at rehearsals or gigs — great for live sound mixing.

ADAT digital I/O
The 16A provides two 8-channel banks of optical digital I/O. Connect outboard digital processors, digital mixers or other gear: 16 channels at 44.1/48 kHz or 8 channels at 88.2/96 kHz. Alternately, the optical Bank A ports can be independently configured to support stereo TOSLink (optical S/PDIF).

Bank A input and output operate independently, allowing you to mix and match optical formats. For example, you could receive four channels of 96 kHz S/MUX input while at the same time sending 96 kHz stereo optical S/PDIF (“TOSLink”) to the output.

Network I/O
AVB stands for *Audio Video Bridging*, a collection of IEEE standards to enable high-bandwidth, low-latency audio streaming over Ethernet. The 16A’s two network ports support 128 channels of network audio input and output for an additional 256 simultaneous audio channels (at 1x or 2x samples rates) using standard CAT-5e or CAT-6 Ethernet cables with lengths up to 100 meters.

Multiple 16A’s or other AVB devices can be daisy-chained up to seven “hops” (links in the chain). Connectivity to a wider network can be achieved with standard AVB Ethernet switches connected to other MOTU AVB devices, 3rd-party AVB devices, and multiple computers, each with full access to all devices on the network (including the other computers).

The entire network operates with near-zero network latency, even over very long cable runs. MOTU’s AVB implementation allows you to stream hundreds of audio channels among devices and computers on the network with guaranteed Quality of Service (QoS), prioritizing audio streams over less important network traffic.

Matrix routing and multing
The 16A provides completely flexible matrix-style audio routing and multing. You can route any analog or digital input, computer channel, or

network stream to any other output, computer, or network device. You can also mult any single input to unlimited multiple output destinations.

Word clock

The 16A supports standard word clock synchronization at any supported sample rate. The word clock OUT port can alternately be used as a THRU port for placing the 16A in a properly-terminated word clock daisy-chain.

Full-color LCD displays

The two 3.9-inch full-color 24-bit RGB IPS TFT LCD displays deliver 480 x 128 pixel resolution (for a combined resolution of 960 x 128) and show all signal activity at a glance with precise, detailed metering for all I/O. You can access many hardware settings directly from the front panel.

Flexible headphone output

The 16A front panel headphone jack includes independent volume control and convenient source signal selection: simply turn the SOURCE knob to choose the signal you wish to hear, as shown in the LCD display. Alternately, you can program the phone output with its own independent mix, or any available source from the 16A's extensive routing matrix.

Control room features

Control room features include talkback and A/B/C monitor select, mute and sum-to-mono for the main outs. Surround monitoring control is also supported for common formats such as 5.1, 7.1.4, and others (up to sixteen channels).

Rack mount or desktop operation

The 16A is housed in a rugged steel full-rack enclosure.

CueMix Pro app

You can control on-board DSP, mixing, device settings, clock/sync settings, and network audio routing from the CueMix Pro app software for macOS, Windows, and iOS. Run CueMix Pro on a tablet or smart phone and control the 16A wirelessly through your local Wi-Fi network. Multiple devices can be used simultaneously to access any audio interface settings on the network.

Stand-alone mixing with wireless control

If you connect your MOTU interface to a Wi-Fi router with a standard Ethernet cable, you can control its powerful mixing and DSP effects from your smart phone or tablet, without a computer — great for live sound mixing from your iPad, tablet, or other wireless device.

Performer Lite

Performer Lite is a full-featured audio workstation software package for Mac and Windows that is available as a free download for you as an 16A owner. Visit motu.com/download to obtain your copy. Performer Lite provides multi-track MIDI and audio production, over 100 included virtual instruments, automated virtual mixing, graphic editing, music notation editing, real-time effects plug-ins with crossfades, support for many third-party audio plug-ins, sample-accurate editing and placement of audio, and more.

CHAPTER 2 Packing List and System Requirements

PACKING LIST

The 16A ships with the items listed below. If any of these items are not present in the box when you first open it, please immediately contact your dealer or MOTU.

- 16A audio interface
- 40 Gbps/240W USB-C cable
- IEC power cord
- Printed Quick Start guide
- Getting Started card

SYSTEM REQUIREMENTS

- Intel Core i3 Mac or PC (or AMD equivalent). Faster CPUs, including Apple silicon, are recommended for best performance.
- 4GB RAM; 8 GB or more recommended.
- Windows 10 22H2 or later; Windows 11 23H2 or later.
- macOS 12 or later. For best possible performance, macOS 13 or later is recommended. See “MacOS compatibility” below for details.
- iOS device running iOS 17 or higher
- Available Thunderbolt, USB4, USB3 or USB2 port.

☛ A Thunderbolt 3 port (or higher) is required for Thunderbolt operation.

- A large hard drive (preferably at least 512 GB).

MACOS COMPATIBILITY

The 16A employs Apple’s latest macOS driver model, which requires macOS 13 or later. Once you run the 16A driver installer on these systems (from the CueMix Pro App *Discovery* tab), the 16A can connect to the host computer via Thunderbolt (128 channels) or USB4 (128 channels) for best possible latency performance.

Using the 16A on macOS 12

The 16A supports USB audio class-compliant operation on macOS 12 (no driver installation needed). Simply connect the 16A to the computer and it will operate as either a USB2 interface (64 channels) or USB3 interface (64 channels), depending on the host computer. Latency performance will be almost as good as Thunderbolt. The CueMix Pro app can be copied to your Applications folder and used as normal.

iOS COMPATIBILITY

The 16A and the CueMix Pro app require an iOS device running iOS 17 or higher. You can use the 16A as an audio interface with your iPhone or iPad using USB class compliant operation. If you have an iPad with an Apple silicon (M series) processor, you can enable the 16A’s low-latency Thunderbolt/ USB driver for best possible performance. See “Software installation for iOS” on page 16.

PLEASE REGISTER TODAY!

Please register the 16A today: *motu.com/register*.

As a registered user, you will be eligible to receive free bundled software, technical support and announcements about product enhancements as soon as they become available. Only registered users can receive the included free software, so please register today.

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

CHAPTER 3 Software Installation

SOFTWARE INSTALLATION FOR macOS

1 Visit *www.motu.com/16A-start* to download the latest macOS *MOTU Pro Audio V2.dmg* virtual disk image.

2 Open the disk image and copy the CueMix Pro app to the Applications folder.

☛ CueMix Pro must reside in the Applications folder for the driver installation process.

3 If you are running macOS 13 or later, launch CueMix Pro. If you are using an earlier macOS version, see “Operation on macOS 11 or 12” on page 16.

4 As CueMix Pro launches, macOS will ask you to allow CueMix Pro to access devices on local networks. It is very important to click *Allow* here.

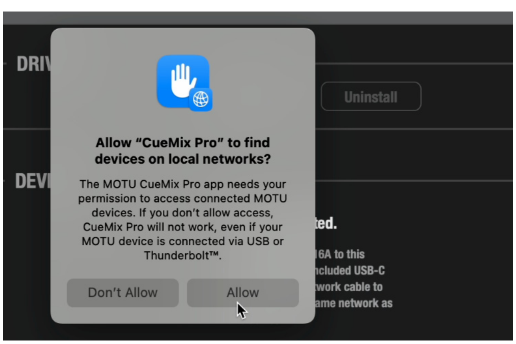


Figure 3-1: Be sure to click ‘Allow’ in this macOS alert, so that CueMix Pro can communicate with your 16A device.

☛ Network access is required for CueMix Pro to communicate with your 16A device, even if it is connected to your computer with Thunderbolt or USB.

5 In CueMix Pro’s *Discovery* tab, click *Install.*

6 When the new driver extension alert appears, click *Open System Settings*.

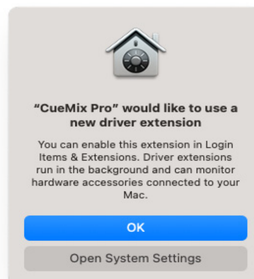


Figure 3-2: Click ‘Open System Settings” in the new driver extension alert.

7 In the resulting *Driver Extensions* dialog, enable the *CueMix Pro* driver extension, as shown in Figure 3-3 (for macOS 15).

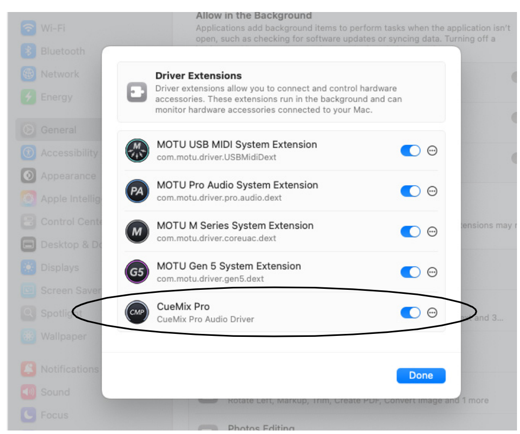


Figure 3-3: Enabling the CueMix Pro Driver Extension (in macOS 15).

Operation on macOS 10.15, 11 or 12

If you are running macOS 10.15, 11 or 12, driver installation (from the CueMix Pro Discovery tab) is not available. Instead, on these older systems the 16A operates as a USB audio class-compliant device that uses the macOS USB audio driver, so no driver installation is needed. You can use CueMix Pro to manage the settings in the interface and use all of its mixing and routing capabilities.

SOFTWARE INSTALLATION FOR WINDOWS

☛ We recommend that you run the software installer *before* you connect the 16A to your computer and power it on. This ensures that all driver components are properly installed in your system.

1 Visit *www.motu.com/16A-start* to download the latest Windows *MOTU Pro Audio v2* installer.

2 Run the installer and follow the directions it gives you.

SOFTWARE INSTALLATION FOR iOS

Audio-class compliant operation allows you to connect the 16A to any iOS device (via either USB-C or a standard Lightning camera connection kit adapter). The 16A then provides multi-channel audio I/O to your iOS audio apps, and you can access all of its settings, features, and mixing with the CueMix Pro app, available for download from the Apple App store.

If you have an iPad with an Apple silicon (M series) processor, you can enable the 16A’s low-latency Thunderbolt/USB driver for best possible performance. Go to *Settings > Apps > CueMix Pro Audio Driver.*

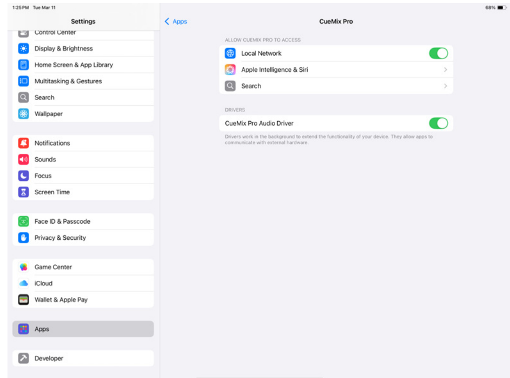


Figure 3-4: Enabling the iPadOS CueMix Pro Audio Driver for lowest possible latency iOS performance. iPads with Apple silicon (M series) processors support the 16A Thunderbolt/USB driver.

AUDIO DRIVERS

The installer provides a Thunderbolt and USB audio driver for macOS 13 and later (CoreAudio) and Windows (ASIO and Wave).

Industry-leading I/O latency performance

On macOS and Windows, the 16A driver provides exceptionally low I/O latency performance. For example, with a 32-sample buffer size, an 16A interface operating at 96 kHz produces round trip latency (RTL) performance of approximately two milliseconds (ms) on macOS and Windows. RTL is the measurement of the time it takes audio to pass from an analog input, through a high-performance DAW host such as Digital Performer, to an analog output.

ASIO Driver support

On Windows, to enable the 16A in your ASIO host software, choose the *MOTU Pro Audio v2* ASIO driver, as shown in Figure 6-1 on page 32.

WDM / Wave driver support

On Windows, the MOTU Pro Audio v2 driver includes support for WDM (Wave) compatible audio software. See item #9 on page 33.

Buffer Size

When connected to a Windows computer, the *Buffer Size* menu is available in the Device tab (item #7 on page 33). This setting determines the amount of latency (delay) you may hear when live audio is patched through your Windows audio software. Smaller buffer sizes produce lower latency, with sizes of 256 samples or less producing virtually imperceptible delay. Many host applications report audio hardware I/O latency, so you can see what happens to the reported latency when making adjustments to this setting.

Be careful with very small buffer sizes, as they can cause performance issues from your host software or PC.

☛ At sea level, audio travels approximately one foot (30 cm) per millisecond. A latency of ten milliseconds is about the same as being ten feet (three meters) from an audio source.

Output Safety Offset

When connected to a Windows host, the *Output Safety Offset* menu in the Device tab of the CueMix Pro app also becomes available (item #8 on page 33). This setting allows you to further reduce host latency. However, the lower latency setting may cause your host software to experience performance issues. Be mindful when choosing the lower latency setting, as this parameter can have a significant impact on your computer system's performance.

CUEMIX PRO APP

CueMix Pro is an easy-to-use app for macOS, Windows and iOS that gives you complete control over all the settings in your 16A interface. For details, see chapter 6, "CueMix Pro" (page 30).

PERFORMER LITE WORKSTATION SOFTWARE

Performer Lite is an easy to use audio workstation software package for macOS and Windows that lets you record, edit, mix, process, bounce and master multi-track recording projects. Advanced features include over 100 included virtual instruments, real-time effects processing, recording, and much more.

To obtain Performer Lite, visit motu.com to register your MOTU audio interface, download Performer Lite and activate it on your computer.

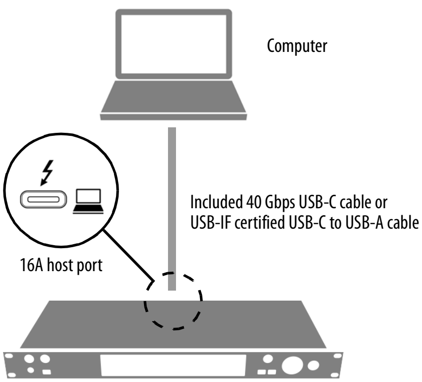
Figure 3-5: Performer Lite.

WORKING WITH HOST AUDIO SOFTWARE

For further information about using the 16A with host audio software, see chapter 7, "Working with Host Audio Software" (page 50).

CHAPTER 4 Hardware Installation

HOST COMPUTER SETUP

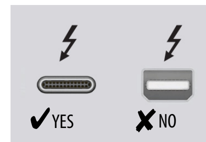


Use this setup if you want to use the 16A as a Thunderbolt or USB audio interface for a computer.

- Use the included 40 Gbps USB-C cable.
- Connect to any USB-C port on the computer or to a USB hub or Thunderbolt dock connected to the computer.

- The 16A negotiates the highest performance connection to the host (Thunderbolt, USB4, USB3 or USB2) based on the what the host supports and the cable used for the connection. The USB-C cable included with the 16A supports all formats. The status section in the front panel LCD (item #12 on page 9) indicates the format of the computer connection achieved.

☛ Thunderbolt operation requires connection to a host computer with a USB-C style Thunderbolt port. Computers with legacy displayport-style Thunderbolt 1 or 2 ports (as shown above) are not supported. *Adapters are also not supported.*



- The 16A does not supply bus power to the host computer, so be sure the computer has its own power source.
- If your computer doesn't have a USB-C port, use a USB C-to-A cable. For best performance and reliability, a USB-IF certified cable is recommended.

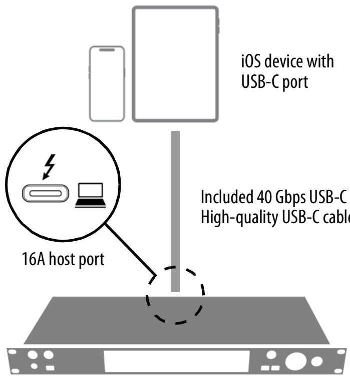
USB-C to USB-A cable

☛ USB-C-to-A plug adapters with a male USB-A plug and a female USB-C port (designed to accept the host-side of the C-to-C cable) *should not be used*, as they are not compliant with USB-IF standards.

This type of USB A-to-C adapter is not recommended.

- For Mac operation, no driver installation is necessary (for USB operation), but it is highly recommended for best performance. Driver Installation is required for Thunderbolt operation. For PC operation, driver installation is required. See chapter 3, "Software Installation" (page 17).

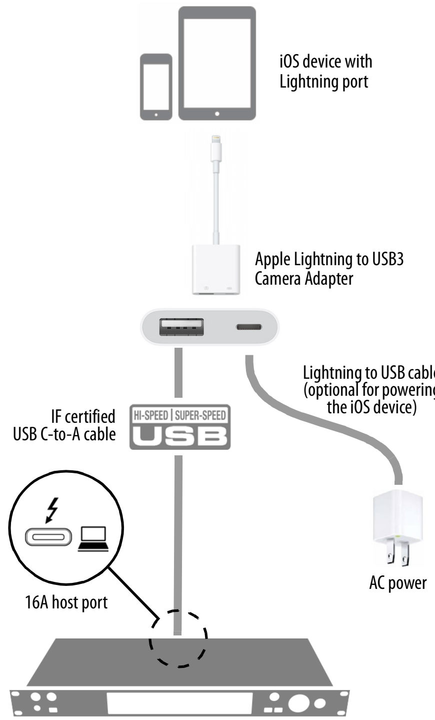
iOS SETUP (USB-C)



Use this setup if you want to use the 16A as an iOS audio interface, or to control it from your iOS device.

- Use this setup for iOS devices with a USB-C port.
- Connect the 16A directly to the iOS device with the included 40 Gbps USB-C cable or a high-quality USB-C-to-C cable.
- For iOS operation, no driver installation is necessary.

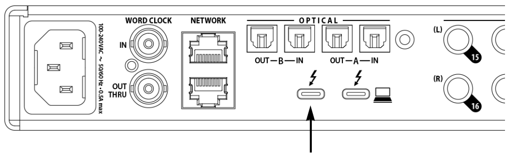
iOS SETUP (LIGHTNING)



Use this setup if you want to use the 16A as an iOS audio interface, or to control it from your iOS device.

- Use this setup for iOS devices with a Lightning port.
- For iOS devices with a Lightning port, an Apple Lightning to USB3 Camera Adapter is required (sold separately), as shown above.
- For iOS operation, no driver installation is necessary.

DEVICE PORT SETUP



The device port allows you to connect additional computer peripheral devices, such as a hard drive or display.

☛ Devices connected to the device port act as peripheral devices for your computer, not the 16A itself. The 16A serves only as a connection between the device and the computer.

What you can connect to the 16A device port depends on the port on the computer that the 16A is plugged into.

☛ Think of the device port as the same as the computer port the 16A is plugged into. If you can successfully connect a device to the computer's port, then you will be able to connect it to the 16A's device port (when the 16A is connected to that same computer port).

Examples of what you can connect to the device port include:

- A hard drive
- A monitor for your computer
- A USB hub
- A Thunderbolt dock

☛ The device port supplies 15W (5V@3A) of bus power for bus-powered devices (USB or Thunderbolt).

☛ Daisy-chained Thunderbolt devices that are bus powered share the 15W power budget supplied by the device port.

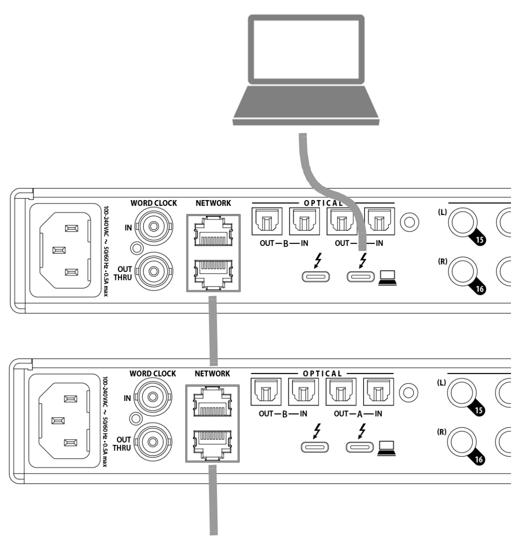
CONNECTING MULTIPLE 16A'S TO A HOST

To connect two or more 16A interfaces to a host computer, use their network ports as discussed in the following sections. For further information, see chapter 9, "Networking" (page 58).

Network daisy-chain

For this approach, connect the first 16A to the host computer (via either Thunderbolt or USB) and then daisy-chain additional units using the two AVB ports as shown below. AVB networking allows seven network "hops" (connections) in a row, which means you can connect up to eight devices in the AVB daisy-chain.

The advantage of this setup is that it can be employed with either Thunderbolt or USB, although Thunderbolt supports more audio channels to and from the host computer (128 in and 128 out). In addition, you do not need to set up an aggregate device in macOS. This setup is also recommended for PC hosts. In this scenario, the 16A units resolve their clocks to one another via their network connections, which prevents their audio streams from drifting apart from one another over time.



Etc. up to eight units total (seven network "hops")

Network switch

A network switch can be used to create a star configuration of interfaces. In general AVB networking provides the most flexible and expandable method for connecting multiple 16A interfaces to a computer (or multiple computers on the network) with long cable runs and very low latency. For further information, see chapter 9, “Networking” (page 58).

A TYPICAL 16A SETUP

See the diagram on page 8 for an example of typical connections to the 16A. The following sections provide important information for achieving best results for each type of connection.

AUDIO CONNECTIONS

Here are a few things to keep in mind as you are making audio connections to your 16A interface.

TRS quarter-inch line inputs

The quarter-inch line inputs are balanced (TRS) connectors. The inputs can also accept an unbalanced (TS) plug.

TRS quarter-inch line outputs

The quarter-inch line outputs are balanced (TRS) connectors. The outputs are DC-coupled, so they can be used for control voltage (CV) output.

☛ The line outputs are not cross-coupled. Therefore, when connecting them to an unbalanced input, use a TRS plug with the ring disconnected. Not floating the negative terminal will short it to the sleeve ground and cause distortion.

Analog I/O calibration

Various settings for the line inputs and outputs, such as gain, trim, phase invert, etc. can be accessed in the Input and Output tabs in the CueMix Pro app. See “Inputs tab” on page 34 and “Outputs tab” on page 35.

All analog inputs and outputs can be calibrated to support a variety of standards, including EBU-R68, SMPTE RP-155, +4dBu, -10dBv, 2vRMS and 1vRMS.

The line inputs are equipped with 0 to +20 dB of digital gain in 1 dB steps.

The line outputs and headphone outs are equipped with a range of digital trim from 0 to -99 dB, adjustable in 1 dB steps.

Monitor A/B/C outputs

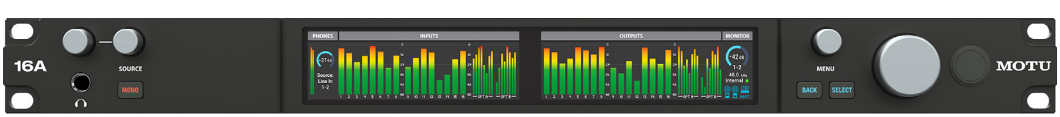
From the factory, Line Outs 1-2 function as monitor pair A. If you have a secondary pair of studio monitors, connect them to Line Outs 3-4 as monitor pair B, as shown in Figure 4-2. A 3rd pair can be connected to Line Outs 5-6 as pair C. You can then use the CueMix Pro app to switch among the three monitor pairs. See “A/B/C Monitoring” on page 48.

Optical I/O

The 16A provides two banks of ADAT optical (“lightpipe”) connectors. Each bank provides an input and output connector. Together, they provide 16 channels of ADAT optical digital I/O at 44.1 or 48 kHz, or 8 channels of SMUX optical at 2x sample rates (88.2 or 96 kHz).

☛ The optical ports are disabled when the interface is operating at a 176.4 or 192 kHz.

Figure 4-1: 16A front panel



TOSLink (optical S/PDIF)
Alternately, the Bank A optical ports can be configured for stereo TOSLink (optical S/PDIF) in the CueMix Pro app (item #4 on page 34 for input and item #9 on page 35 for output). The optical IN and OUT banks can be configured independently.

Choosing a clock source for optical connections
When connecting an *optical* device, make sure that its digital audio clock is rate-locked (in sync with) the 16A, as explained in “Synchronization” on page 22 and “Syncing optical devices” on page 23.

SYNCHRONIZATION
If you connect devices digitally to the 16A, or if you need to synchronize the 16A with an outside time reference such as word clock, you must pay careful attention to the synchronization connections and clock source issues discussed in the next few sections.

Do you need to sync?
If you will be using only the 16A’s analog inputs and outputs (and none of its digital I/O), and you don’t need to resolve your system to external word clock, you don’t need to make any sync connections. You can skip this section.

Situations that require synchronization
There are two general cases in which you will need to resolve the 16A with other devices:

- Synchronizing with other digital audio devices so that their digital audio clocks are *rate-locked*
- Resolving the 16A to an external clock source

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 digital I/O features on the 16A 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 rate-locked with one another, so that their clock rates don’t drift with respect to one another over time. Otherwise, you’ll hear clicks, pops, and distortion in the audio — or perhaps no audio at all.

There are two ways to achieve rate 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.

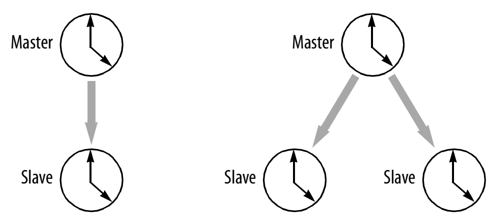
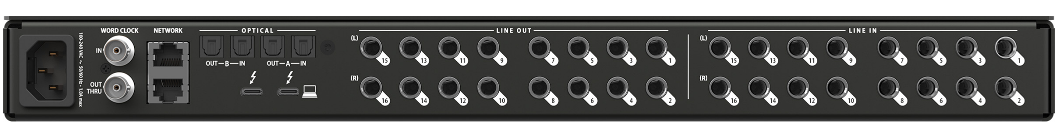


Figure 4-3: To keep the 16A rate-locked with other digital audio devices connected to it, choose a clock master.

Also remember that audio rate 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

Figure 4-2: 16A back panel



you set things up with this rule in mind, you'll have trouble-free audio transfers with your MOTU hardware.

SYNCING OPTICAL DEVICES

There are several ways to sync an optical device with the 16A:

A. Resolve the other device to the 16A

B. Resolve the 16A to the other device

C. Resolve both devices to a word clock source

For A, choose *Internal* as the clock mode in the Device tab (item #5 on page 33). Then configure the other device to resolve to its optical input. Alternately, the MOTU interface could be resolved to any other external clock source besides *Optical*.

For B, choose *Optical* as the clock mode (item #5 on page 33), and configure the other device to resolve to its own internal clock.

For C, choose *Word Clock* as the clock mode for the 16A (item #5 on page 33), and resolve the other device to its word clock input.

Using word clock to resolve optical devices

If the optical device has word clock connectors on it, you can use them to synchronize the device with the 16A. See the next section, "Syncing word clock devices".

SYNCING WORD CLOCK DEVICES

The word clock connectors on the 16A 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 22). 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 4-4 and Figure 4-5.

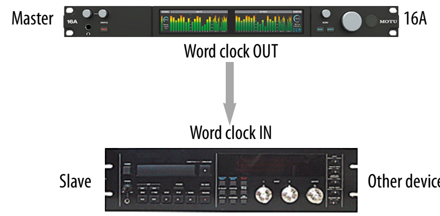


Figure 4-4: Slaving another digital audio device to the 16A via word clock. For the 16A clock source, choose Internal (or any source other than Word Clock, as daisy-chaining word clock is not recommended).

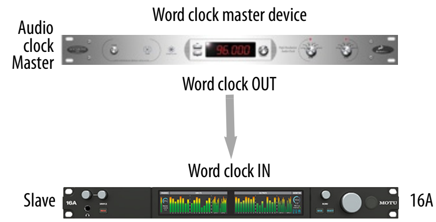


Figure 4-5: Slaving the 16A to word clock. For the 16A clock source, choose 'Word Clock In'.

The 16A word clock input provides proper 75 ohm termination, but only when the BNC jack is configured to operate as a word clock output. Alternately, the word clock output can be configured as a word clock thru, which connects the input clock signal directly to the output, *with no termination on the input*. See the "Daisy-chaining word clock" below.

Resolving to a word clock signal that matches the 16A base clock rate

The 16A can resolve to a word clock signal running at an even multiple of the current system clock setting (the *base clock rate*). For example, the 16A could be running at 96 kHz while resolving to a 48 kHz word clock signal from another device. Similarly, the 16A could run at 88.2 kHz and resolve to 44.1 kHz word clock signal. Conversely, the 16A could run at 48 kHz and resolve to a 96 kHz word clock signal. However, if the 16A is running at 96 kHz, it cannot resolve to a word clock signal running at 44.1 kHz.

In summary, the word clock signal must be one of the following:

- the same as the current 16A clock rate
- 2x or 4x the current 16A clock rate
- half or quarter of the current 16A clock rate

Daisy-chaining word clock

If necessary, you can daisy-chain several word clock devices together. When doing so, connect WORD CLOCK OUT from the first (master) device to the WORD CLOCK IN on the second device. Then connect its WORD CLOCK THRU port to the next device's WORD CLOCK IN port, and so on. If the 16A is in the middle of the chain, use its WORD CLOCK OUT port and change its operation from OUT to THRU (using item #6 on page 33 or from the front panel menu). If the 16A is the first device in the chain (to generate word clock for the other devices), do not enable WORD CLOCK THRU mode.

Make sure that the last device in the word clock chain has 75 ohm termination on its input.

If you have more than four word clock devices that you need to synchronize, try to avoid chaining their word clock connections. Instead, use a word clock distribution device of some kind.

☛ When the 16A is powered off, the WORD CLOCK OUT reverts to THRU.

SYNCING AVB DEVICES

When connecting AVB streams between two or more devices on a network, one device needs to be designated as the clock master, while all other devices use it as a clock source. Set the master device's Clock Source to *Internal*. Other devices should then resolve to the master device via an AVB stream connection.

For details about how to set this up between multiple 16A's, or between the 16A and other AVB devices, see "Syncing AVB devices" on page 63

Part 2
Using the 16A

CHAPTER 5 Front Panel Operation

The front panel provides access to basic settings and the headphone output with independent volume and source select. The high-resolution LCD screens display level meters for all inputs and outputs, status information and activity indicators for network activity. Basic device settings and status information can be accessed in the LCD using the MENU knob and SELECT/BACK buttons.

LEVEL METERS

In its default state when the unit is first powered on, the two LCD screens display level meter activity for all audio inputs and outputs (Figure 5-2).

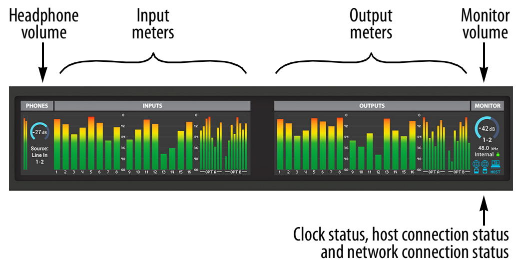


Figure 5-2: The front panel displays.

The Status section

The Status section (Figure 5-3) displays basic information about the 16A.

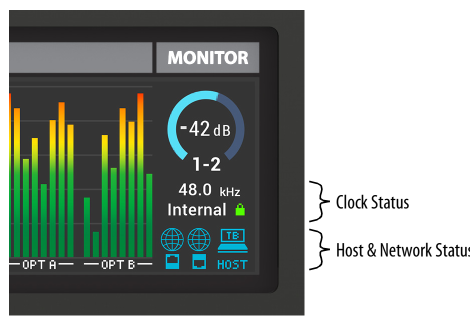


Figure 5-3: The Status section.

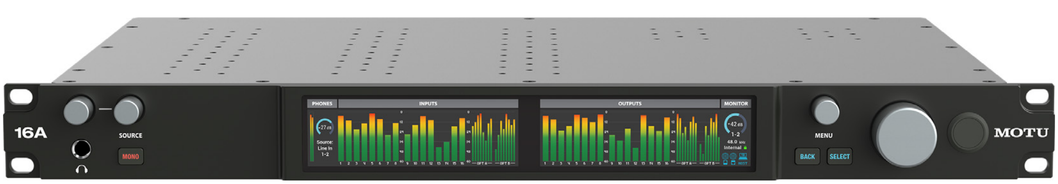


Figure 5-1: The 16A front panel.

Clock Status

The Clock Status section of the screen (Figure 5-3) displays the *Sample Rate* at which the unit is currently operating, and the current *Clock Source* setting (item #6 in the Devices tab on page 33). The Sample Rate and Clock Source settings can also be found (and changed) in the front panel screen menu. They can also be changed in the Device tab of CueMix Pro (page 33).

The Lock icon

When the 16A has successfully resolved to the current clock source, the *Lock* icon (Figure 5-3) turns *green*. When the 16A has not yet successfully locked to the current clock source for some reason, the lock icon turns *red*. Check the *Clock Source* setting (item #6 on page 33), cable connections, etc.

HOST Status

The *Host* icon (Figure 5-3) indicates the status of the USB connection to the computer, as shown below. *Blue* indicates a successful connection to the host; the number (or letters) indicates the format of the connection (Thunderbolt/USB4, USB3 or USB2). When the icon turns *yellow*, this indicates that a connection has been detected, but full connectivity has not yet been establish for some reason.

Host connection status	Host icon	
No connection		(White)
Connection detected, but not yet fully established		(Yellow)
Thunderbolt/USB4 (PCIe)		(Blue)
USB3		(Blue)
USB2		(Blue)

Network Status

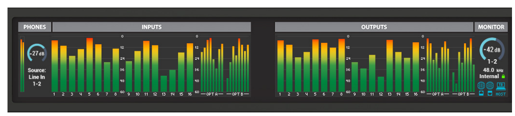
The *Network Status* indicators (Figure 5-3) turn blue when a network connection has been successfully established on the top or bottom network port on the rear panel of the 16A.

Network connection status	Network icon	
No connection		(White)
Connection established		(blue)

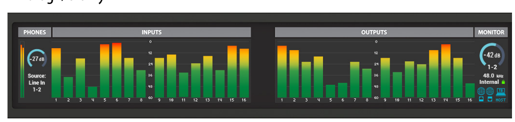
Meter views

Several alternative meter views are available, such as *All meters* and *Analog I/O only*. These can be chosen from the MENU (see the next section).

All meters



Analog I/O only



Digital I/O only

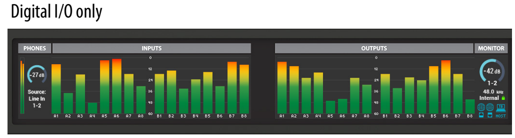


Figure 5-4: Meter views.

MENU NAVIGATION

Turn the MENU knob or push the SELECT button (Figure 5-1) to access the menu, which provides settings and status information. Turn the MENU knob to scroll through the menu settings.

Push SELECT to enter the selected sub-menu or to select the currently highlighted parameter. Push BACK to go to the parent menu.

To exit the menu entirely, push BACK repeatedly until the menu disappears from the display.

Menu Item	What it does
Sample Rate	Sets the sample rate for the device.
Clock Source	Sets the digital audio clock source for the device. See “Synchronization” on page 22.
Word Clock	Lets you configure the Word Clock OUT port as either OUT or THRU. See “Syncing word clock devices” on page 23.
Optical A Format	Specifies ADAT or TOSLink. See “Optical I/O” on page 21.
Optical Expander Presets	Provides presets for optical expander operation at each sample rate (up to 96 kHz). See chapter 8, “Optical Expander Presets” (page 56).
Meter View	Lets you choose a desired metering configuration, such as *All Meters*, *Analog I/O Only*, etc.
Meter Settings	Lets you set the peak/hold time for the meters.
Display	Provides timeout options for the display (after 30 seconds, 30 minutes or off). When timed out, the screen goes dark.
Network	Displays the IP address and network clock source for the 16A unit.
About	Displays the device name, serial number, and firmware version for the device.
Presets	Lets you save and recall device presets, which store the entire state of the interface. Up to eight different presets can be stored.
Reset	Restores factory default settings. Please note: doing a factory reset will erase any saved user presets. To restore factory settings without erasing saved user presets, see “The Default Preset” on page 29.

HEADPHONE VOLUME

Turn the headphone volume knob (Figure 5-1) to adjust the phones volume. The LCD indicates the current level (on a scale from -∞ to 0) and provides feedback during adjustment.

HEADPHONE SOURCE

Turn the headphone SOURCE knob (Figure 5-1) to choose the channel you wish to listen to on the headphones. You can either wait a brief moment for your selection to take effect, or you can push SELECT to confirm your choice (or BACK to cancel it).

The MONO button

When choosing the headphone source signal, the menu displays stereo pairs (e.g. Line Out 1-2, Line Out 3-4, etc.) If you would like to choose just a single channel (e.g. Line Out 7), press the MONO button. Now, the SOURCE knob scrolls through individual channels. The mono signal from the chosen channel is split and sent to both left and right channels on your headphones. Press MONO again to turn off mono mode and return to stereo operation.

MONITOR CONTROL

Turn the large volume knob (to the right of the LCD) to adjust the Monitor volume. From the factory, this knob controls the level of Line Outs 1-2 (and Line Outs 3-6 when operating in A/B/C mode). If you add additional outputs to the Monitor group (see below), this volume knob controls the output level of all of them.

THE MONITOR GROUP

The 16A *Monitor Group* allows you to add additional analog outputs to be controlled as a group by the main volume knob (and the corresponding volume control in the Home tab and Output tab in CueMix Pro). For example, you could group Outputs 1-6 or 1-8 to control a 5.1 or 7.1 surround system, respectively. Choose the desired outputs in the *Monitor Group* section in the CueMix Pro Home tab (page 32) or Output tab (page 35). After doing so, the main volume knob (on both the front panel and in the CueMix Pro Home and Output tabs) affect all of them.

When A/B/C Mode is enabled, the Monitor Group controls all three pairs of A, B and C outputs (Line Out 1-2. Line Out 3-4, and Line Out 5-6, respectively). See “A/B/C Monitoring” on page 48. Use the trim controls in the CueMix Pro Output tab to trim Monitor Group outputs relative to each other.

SAVING AND RECALLING DEVICE PRESETS

Use the *Presets* menu command on the front panel of the 16A to save and recall up to eight 16A device presets in the device. A preset saves the entire state of the 16A interface, including all device settings, mixes and effects settings.

The Default Preset

The *Default Preset* restores the factory default settings of the unit while at the same time preserving any other presets you may have saved in the other preset slots.

POWER BUTTON

Push the power button to turn the unit on or off.

CHAPTER 6 CueMix Pro

CueMix Pro gives you complete control of all the settings in the 16A. It is a standard software application installed on your Mac or PC when you run the MOTU Pro Audio V2 installer or setup app. It can be found in the Applications folder (Mac) or Start menu under MOTU (Windows).

CueMix Pro is also available as an iOS app.

RUN THE INSTALLER, GET THE APP

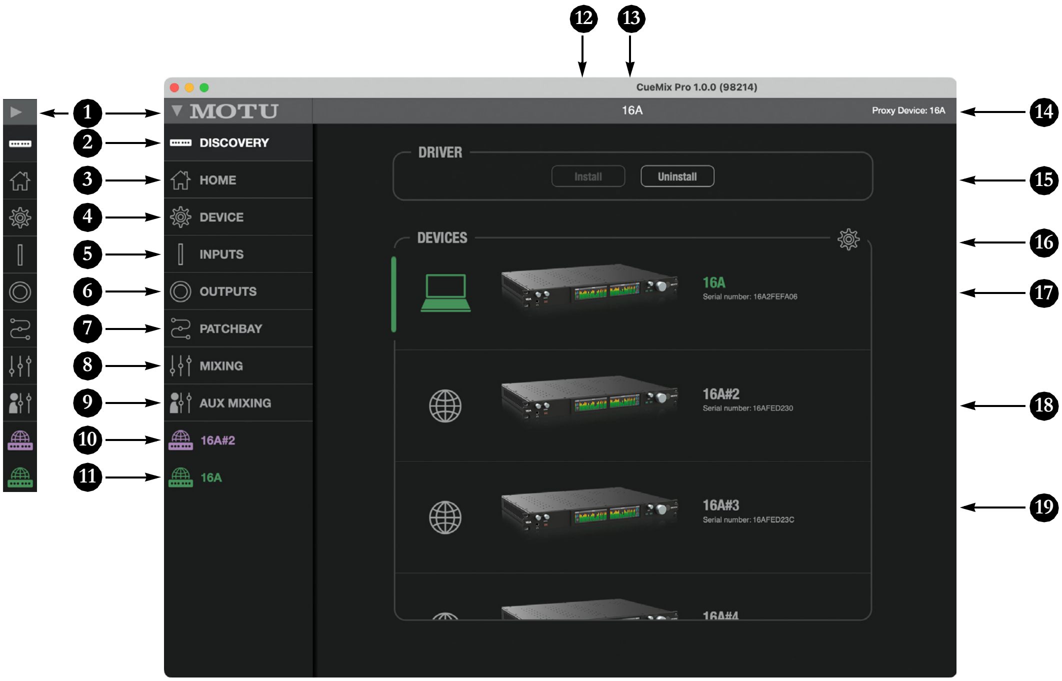
Visit *motu.com/16A-start* to get the latest MOTU Pro Audio v2 installer or setup and run it on your computer. Visit the Apple App Store to install the CueMix Pro app on your iOS device.

☛ Look for updated PDF versions of this user guide at the link above, which may document new features and updates to CueMix Pro.

MAKE HARDWARE CONNECTIONS

Before running CueMix Pro, be sure that the 16A is successfully connected to your device and powered on, and described in chapter 4, “Hardware Installation” (page 18).

DISCOVERY TAB



1. Expands and collapses the sidebar.

2. This is the *Discovery* tab. It lets you access and manage all 16A's connected to your computer and local area network (LAN).

3. The *Home* tab provides quick access to basic settings. See "Home tab" on page 32.

4. The *Device* tab provides basic hardware settings, such as the Sample Rate and Clock Source. See "Device tab" on page 33.

5. The *Inputs* tab provides settings for the 16A's physical inputs, such as gain levels for the line inputs. See "Inputs tab" on page 34.

6. The *Outputs* tab provides settings for the 16A's physical outputs, such as trim levels for the line outputs. See "Outputs tab" on page 35.

7. The *Patchbay* tab provides flexible audio channel patching and routing among sources and destinations, including the computer, the 16A physical inputs and outputs, the 64-channel mixer in the 16A, and AVB network I/O streams.

8. The *Mixing* tab provides access to the on-board mixing and effects. The 16A is a capable 64 x 32 monitor mixer. See "Mixing tab" on page 37.

9. The *Aux Mixing* tab gives you access to send faders for each aux mix. See "Aux Mixing tab" on page 38.

10. This area at the bottom of the sidebar displays any AVB devices on the same network as the currently selected (green) device here in the Discovery tab (17), which serves as the network *proxy device* (14). The sidebar list shows both MOTU and 3rd-party devices. Click a device to view its settings in CueMix Pro. When previous-generation MOTU devices or 3rd-party devices are selected, they will display only basic settings in CueMix Pro. Unavailable tabs will be grayed out.

11. This is the currently selected device (highlighted in green). This is the device currently being viewed in the CueMix Pro app.

12. The *Devices* list shows any 16A's detected by CueMix Pro. In the future, other similar products in the 16A (Pro Audio v2) family will appear in this list as well, when detected by CueMix Pro.

13. The title bar displays the name of the 16A or other device on the network that is currently being viewed in (and controlled with) CueMix Pro.

14. The *Proxy Device* provides access to any other AVB devices on the network. These devices are displayed in the sidebar (10).

15. (macOS only) The *Driver* section provides buttons for installing and uninstalling the 16A driver, which supports the best possible performance for your 16A.

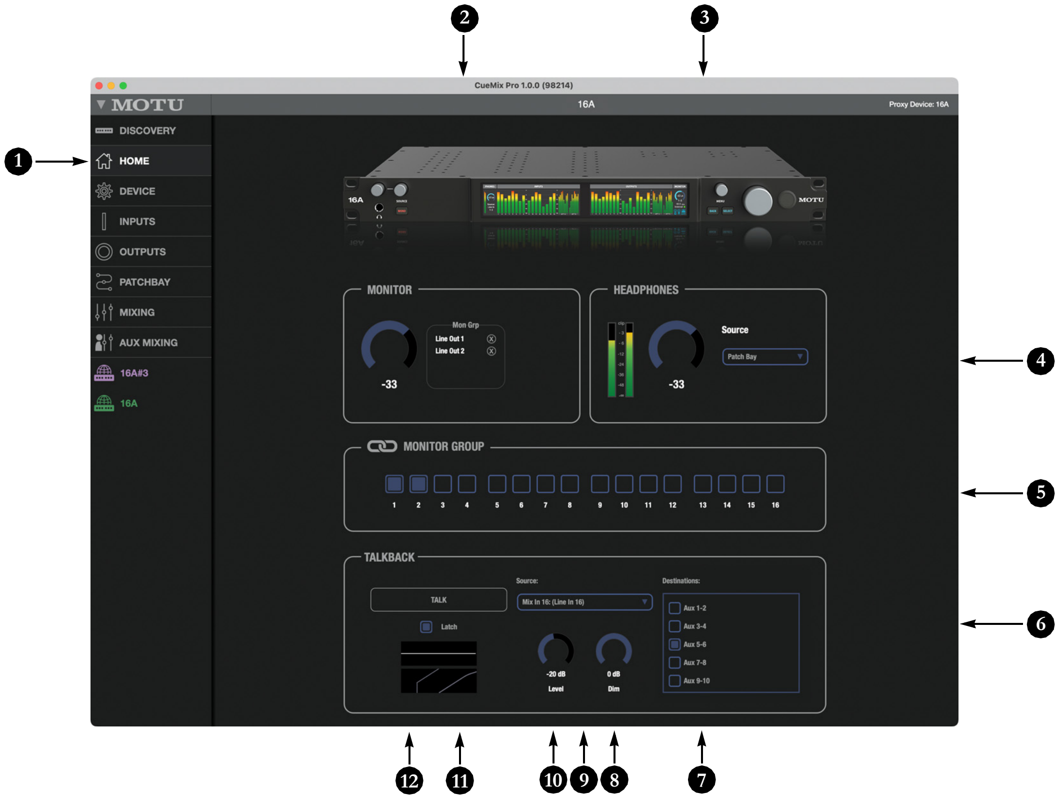
16. This menu lets you show or hide a virtual 16A in the device list. Clicking on the virtual 16A lets you work with CueMix Pro off line, when no 16A hardware is accessible or available (i.e. there is no 16A present in the list of available devices). You can access all CueMix Pro settings in all tabs.

17. The *currently selected device* is displayed in green. Click a device to select it. This is the device that CueMix Pro is connected to. When you connect to a device here, you can view its settings in CueMix Pro. It also serves as the *Proxy Device* (14), providing access to all other AVB devices on the same network; network devices are displayed in the sidebar (10).

18. Each devices displays its connection type (host or network).

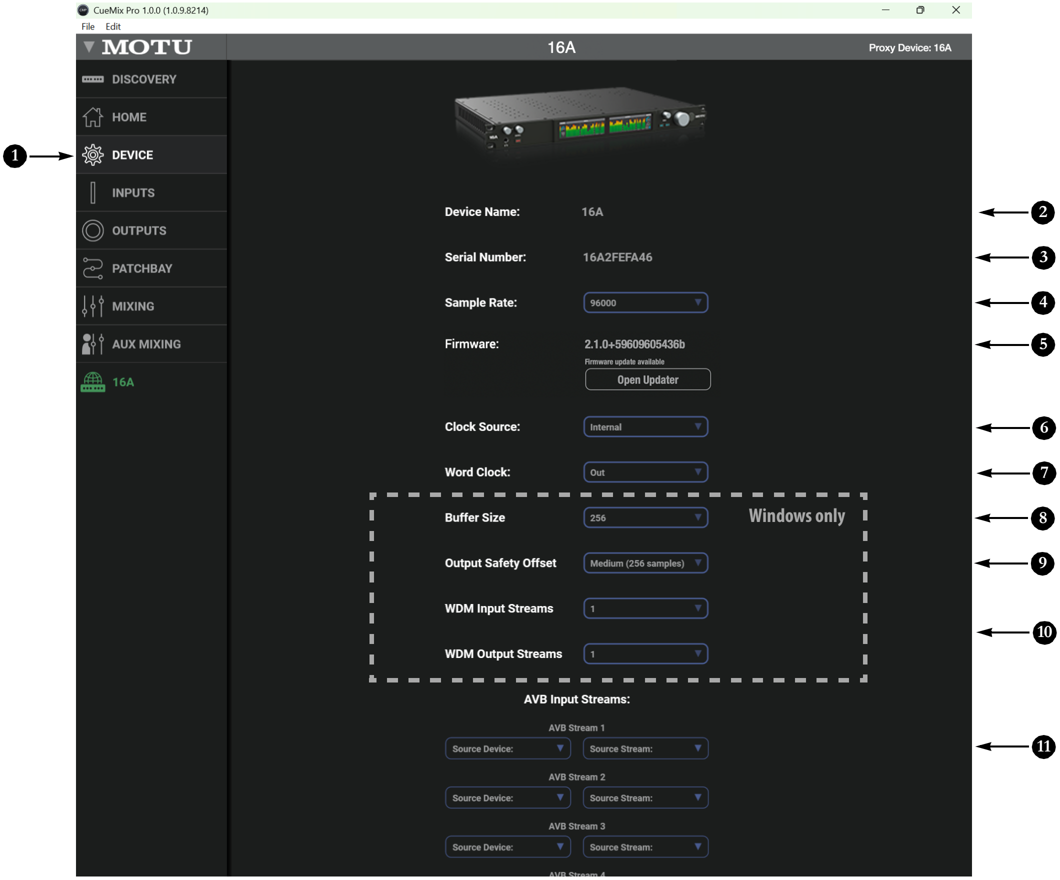
19. To rename a device, click it to select it and then go to the Device tab (4) to rename it there.

HOME TAB



1. This is the *Home* tab, which provides quick access to basic settings.
2. Control the unit's *Monitor* volume here (the level of the Monitor Group). This is the same as the main volume knob on the front panel of the unit (item 8 on page 9).
3. Control the unit's *Headphones* volume here. This is the same as the headphone volume knob on the front panel of the unit (item 1 on page 9).
4. Choose the *Source* signal for the headphone output from this menu. This is the same setting controlled by the SOURCE knob on the front panel of the unit (item 2 on page 9).
5. The *Monitor Group* represents analog outputs that are dedicated to monitoring in your studio. From the factory, the Monitor Group includes Analog Outs 1-2, which you would connect to your primary studio monitors. You can add additional outputs for multi-channel monitoring. All outputs in the Monitor Group are controlled by the main volume knob on the front panel of the 16A, and the Monitor Volume knob in this tab (2). You can adjust their relative volume to one another using the trims in the Output tab (page 35).
6. The *Talkback* section provides controls for setting up and controlling the 16A's talkback features. See See "Talkback" on page 48.
7. Use mixer buses to route the talkback signal to the desired destination in your studio. Select the bus you are using here in the *Destinations* list.
8. When engaged, Talkback can temporarily reduce the level of all other signals on the talkback bus. Use the *Dim* control to determine the signal reduction amount.
9. Use the *Source* menu to choose the source signal for talkback, such as an analog input connected to a talkback mic.
10. *Level* controls the volume of the Talkback signal.
11. Click the EQ and dynamics processing thumbnails to access a 4-band EQ, gate and compressor for the talkback signal.
12. Click the *TALK* button to engage talkback. If the *Latch* button is disabled, the TALK button will only engage while pressing on the button. With Latch enabled, you can click the TALK button and it will remain on until you click it again.

DEVICE TAB



1. This is the *Device* tab, which provides basic hardware settings, such as the Sample Rate and Clock Source.

2. You can give your 16A a unique *Device Name*, which appears in the Discovery tab, which allows you to manage multiple devices that are connected or available on the network.

3. Displays the *Serial Number* of your 16A device.

4. Choose the desired *Sample Rate*. Make sure your host audio software is set to the same rate.

5. Displays the *firmware version* currently installed in your 16A device. If an update is available, click *Open Updater*. See "Firmware Updater" on page 49. Be sure to check frequently for updates for new features and enhancements.

6. Choose the *Clock Source*. Your MOTU device will resolve its digital clock to this master source. Set the clock source to *Internal*, unless you have other devices connected to the optical inputs. If so, see "Optical I/O" on page 21. If you are resolving the 16A to an external word clock source, choose *Word Clock*. See "Syncing word clock devices" on page 23. If you are working with other devices on an AVB network, you can also resolve the 16A to an AVB stream connection. See "Syncing AVB devices" on page 63.

7. The *Word Clock* output on your MOTU interface can operate as an OUT or a THRU. When Word Clock *Thru* is enabled, 75 ohm termination on the input is lifted and the word clock signal received on the input is patched directly to the Word Clock Output. See "Daisy-chaining word clock" on page 24.

8. (Windows only) Choose the desired *Buffer Size*. Smaller values reduce latency but increase your computer's CPU load. See "Buffer Size" on page 17.

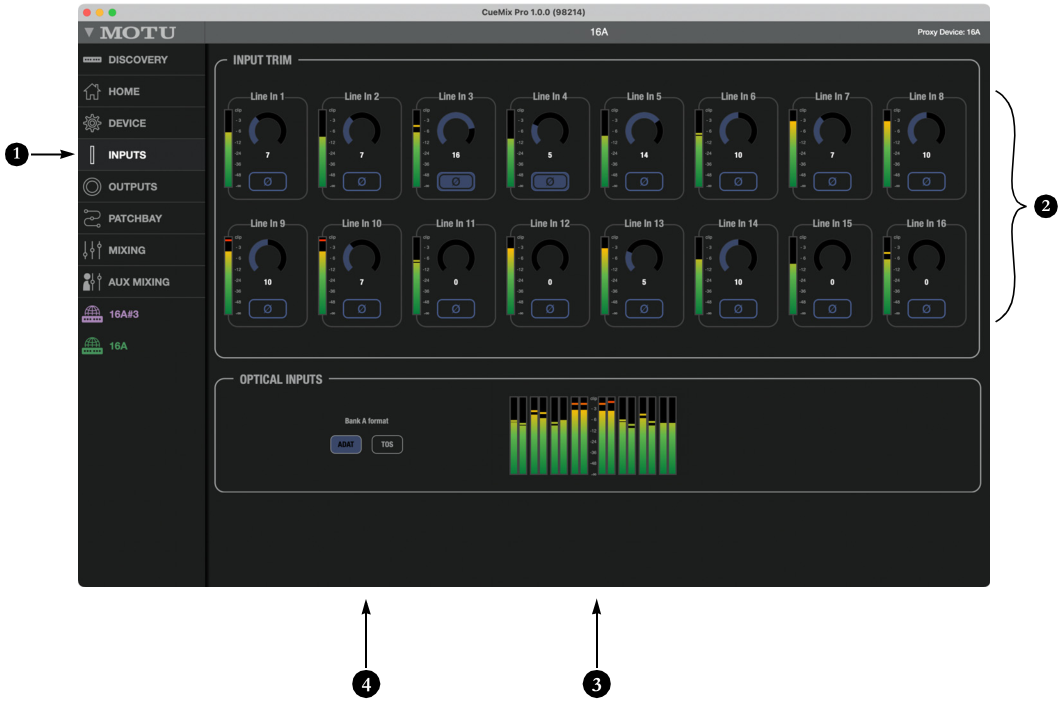
9. (Windows only) Use the *Output Safety Offset* setting to fine tune host buffer latency. See "Output Safety Offset" on page 17.

10. (Windows only) The 16A supports Windows built-in audio. Choose the number of *WDM Input Streams* and *WDM Output Streams* you wish to use with your Windows audio applications that use built-in audio. These settings do not affect the ASIO driver channels.

11. The 16A provides sixteen *AVB Input Streams* for receiving audio from other AVB devices on the network. Each stream supports from one to eight audio channels. Use these menus to select the *Source Device* and *Source Stream* for each 16A input stream. You can then use the Patchbay (page 36) to route the input streams elsewhere in the 16A (to the host computer, the built-in 16A mixer, 16A outputs, etc.) See "Setting up the 16A for networking" on page 61.

12. (Not shown) The 16A also provides sixteen *AVB Output Streams* (scroll past the Input Streams section to access them). Output streams allow you to send 16A audio signals to other devices on the AVB network. Here in the Device tab, you can configure the number of audio channels for each output stream, and choose the stream format. See "AVB output stream format" on page 62 and "Setting up the 16A for networking" on page 61.

INPUTS TAB



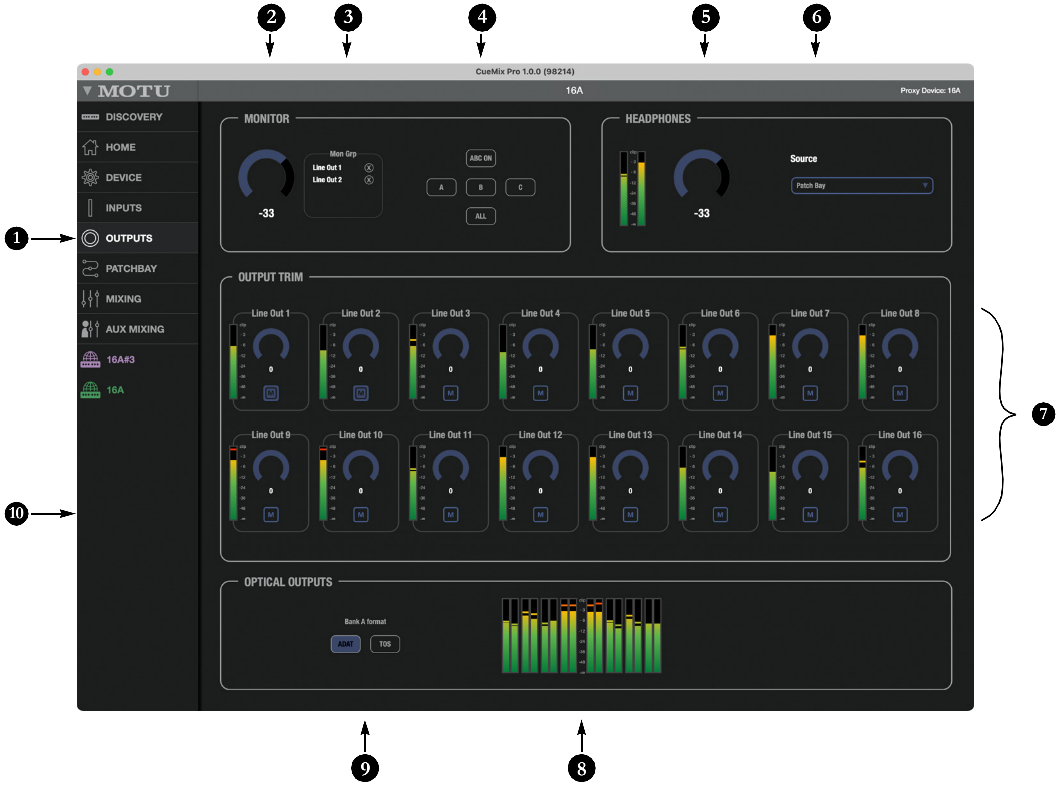
1. This is the *Input* tab, which provides access to settings for the 16A analog and digital (optical) inputs.

2. Each line input can be digitally boosted up to + 20 dB. This allows the inputs to easily accommodate +4 dB and -10 dB reference levels. Each input also includes a *Phase Invert* button.

3. Meters are provided here for the two Optical Input banks, for convenience.

4. Configure the *Optical Input format* for Bank A for either 8-channel ADAT or stereo TOSLink. At 88.2 or 96 kHz, the ADAT setting supports 4-channel SMUX format. Note that you can choose a different format for the optical IN and OUT ports. See “Optical I/O” on page 21.

OUTPUTS TAB



1. This is the *Output* tab, which provides settings for the 16A's analog and digital outputs.

2. MONITOR volume controls the level of the monitor group. This is the same as the main volume knob on the front panel of the 16A and in the Home tab (item #2 on page 32).

3. The *Monitor Group* determines which outputs are controlled by the *Main* volume knob on the 16A's front panel (item #8 on page 9), plus the Main Volume controls in the Home tab (item #2 on page 32) and here in the Output tab. For example, if your studio has a pair of main monitors, plus a sub-woofer connected to Line Out 3, you could click the 'M' button for *Analog 3* to add it to the main volume controls. If you have 5.1 or 7.1 surround monitoring, you can add the surround channels to the monitor group to be able to control the volume of all surround outputs simultaneously with the Main Volume knob.

4. *A/B/C Monitoring* lets you connect two or three sets of speakers and then check your mixes on each pair by switching among them using the A, B and C buttons. See "A/B/C Monitoring" on page 48.

5. Meters and volume controls for the *Headphone* output. These are the same as the headphone volume control on the front panel.

6. Choose the *Source* signal for the headphone output from this menu. This is the same setting controlled by the SOURCE knob on the front panel of the unit.

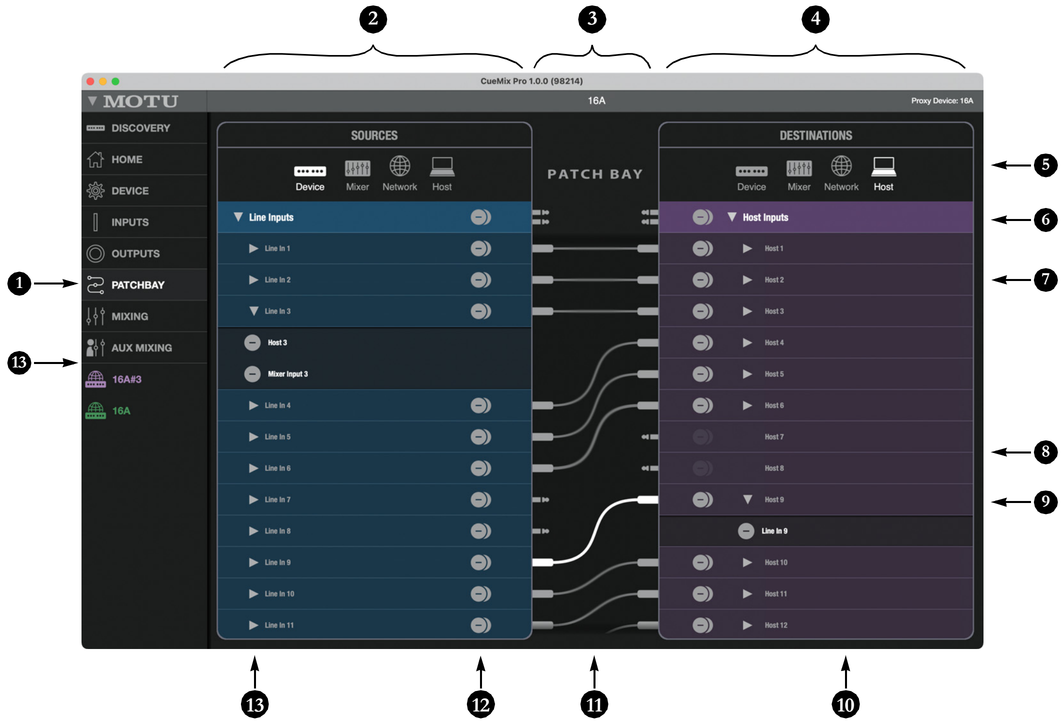
7. All analog outs can be *trimmed* from zero to -∞ dB. This can be useful for speaker calibration or other situations where you need a fixed amount of level adjustment for a particular output (or output pair).

8. Meters are provided here for the two Optical Input banks, for convenience.

9. Configure the *Bank A format* for either 8-channel ADAT or stereo TOSLink. At 88.2 or 96 kHz, the ADAT setting supports 4-channel SMUX format. Note that you can choose a different format for the IN and OUT. See "Optical I/O" on page 21.

10. Enable the 'M' button to include the output in the *Monitor Group*. The Monitor Group represents analog outputs that are dedicated to monitoring in your studio. From the factory, the Monitor Group consists of Analog Outs 1-2, which you would connect to your primary studio monitors. You can add additional outputs for multi-channel monitoring. All outputs in the Monitor Group are controlled by the main volume knob on the front panel of the 16A, the Monitor Volume knob in this tab (2), and the Home tab (item #2 on page 32). You can adjust their relative volume to one another using the trims (7).

PATCHBAY TAB



1. This is the *Patchbay* tab, which lets you make connections between audio sources and destinations in your 16A system.

2. This bank on the left represents *Sources*, which are presented in four separate sub-banks:

   Device — Analog and digital (optical) inputs on the 16A device itself.

   Mixer — Bus outputs from the 16A's built-in mixer, which has 32 output buses.

   Network — Incoming AVB audio streams from other devices on the network.

   Host — Thunderbolt or USB audio channels from the host computer.

   Channels are grouped into banks that can be collapsed or expanded.

3. This is the *Patchbay*, where you drag cables from one bank to the other to make connections, just like a real patchbay. Grab a cable from either side and snap it into any desired socket on the other side. To grab several cables at once, press and hold your mouse on the first one and then drag down or up to select additional adjacent cables; then drag them together. To grab an entire bank, such as all *Analog Inputs*, drag its "double" cables next to the bank name. To clear a connection, click it to select it and hit the delete key. To select multiple connections, drag across them.

   Sources can be connected to multiple destinations, but destinations can only have one source.

   When making or clearing connections, you can use *Undo* and *Redo* in the Edit menu.

4. This bank on the right represents *Destinations*, which are presented in four separate sub-banks:

   Device — Analog and digital (optical) outputs on the 16A device.

   Mixer — Inputs to the 16A's built-in 64-channel mixer.

   Network — Outgoing AVB audio streams to other devices on the network.

   Host — Thunderbolt or USB audio channels to the host computer.

5. Click the desired source or destination bank here to view it.

6. This is the *Computer Inputs* bank, which represents Thunderbolt or USB audio channels going to the host computer. It is currently expanded so you can see individual channels below. The "double cable" in the Patchbay lets you connect all channels in this bank in one operation.

7. Here, the 16A's Analog Line Input 2 is connected to host audio channel 2. This connection routes audio from Line Input 2 to Thunderbolt (or USB) input channel 2 in your host software.

8. Here, host input channels 7-8 are not connected to any 16A sources yet. Grab their cable icon and drag it to the desired source.

9. Expand a channel to see what it is connected to. Click the minus (-) button to clear the connection. To clear all connections to/from a source or destination, click its "double" minus (-) button.

10. Drag vertically on a bank to scroll it, or use the scroll wheel on your mouse (or your finger on iOS).

11. Scroll either bank vertically to see more connections. The white connection is selected; hit the delete key to clear it.

12. Click the double-minus (-) button to clear all connections from the source or destination.

13. Use the expand buttons to see what the source is connected to. Note that sources can be connected to multiple destinations, either in the same bank or other banks.

14. In this example, analog (line) input 2 is being patched to *Host Channel 3* and *Mixer Input 3*. The mixer connection is not currently shown, but if you switch the destination bank to mixer (on the right) you'll see it then.

MIXING TAB

1. The *Mixing tab* gives you access to the 16A's 64-channel 32-bus mixer. The mixer provides 64 input channel strips, followed by 32 bus faders. Scroll horizontally to view them. The current bus fader and its channel strip (*Main 1-2* by default) stays pinned to the right side of the window. It can be changed to any bus using the *On Faders* setting (15).

2. Click the *Mix Settings* menu icon to open (or close) this sidebar. The top three items (*Channels*, *Buses*, and *Sections*) let you show and hide channels strips and sections. For example, you can hide inputs you are not using, or the Compressor/ Limiter thumbnails. Click their disclosure triangle to view each list.

3. This is an *input channel strip*. For details, see "Input channel strips" on page 39. There are 64 input channels, which you can show or hide using the *Channels* sidebar section (2). If you scroll to the right of the input channels, you'll see 32 bus faders, including the reverb bus, 26 aux busses, the *Monitor* bus, and the *Main 1-2* bus. For details, see the sections later in this chapter.

4. This output bus channel strip is always pinned to the right side of the window. By default, it shows the *Main 1-2* bus channel strip, but it can be changed to any bus using the *On Faders* setting (15).

5. For input channels, this displays the source signal for the input. For bus channel strips, this shows the name of the bus. Click it to access basic settings for the channel or bus.

6. Thumbnail for the *4-band EQ* for the channel or bus. Click to access the graphic EQ controls.

7. Thumbnail for *dynamics processing* for the channel or bus. Inputs have a gate and both inputs and busses have a compressor. Click to access graphic controls.

8. *Send fader* thumbnail that shows any send faders that are up for the input. Click to access all send faders for the input, consolidated in one window. See "Mix Sends" on page 44.

9. *Pan control* for mono input channels. Inputs that are grouped as stereo pairs do not have this pan control. To create stereo pairs, click the channel settings (5) for one of the two channels or any thumbnail.

10. *Mute* the input or bus here. Glide horizontally across multiple channels to mute them in one gesture.

11. *Channel* or *bus fader* and *level meter*. The fader range is from -∞ dB to +12 dB. Double-click the fader to jump to unity gain or -∞. Level meters are prefader.

12. *Fader level* in dB. Click to edit it numerically.

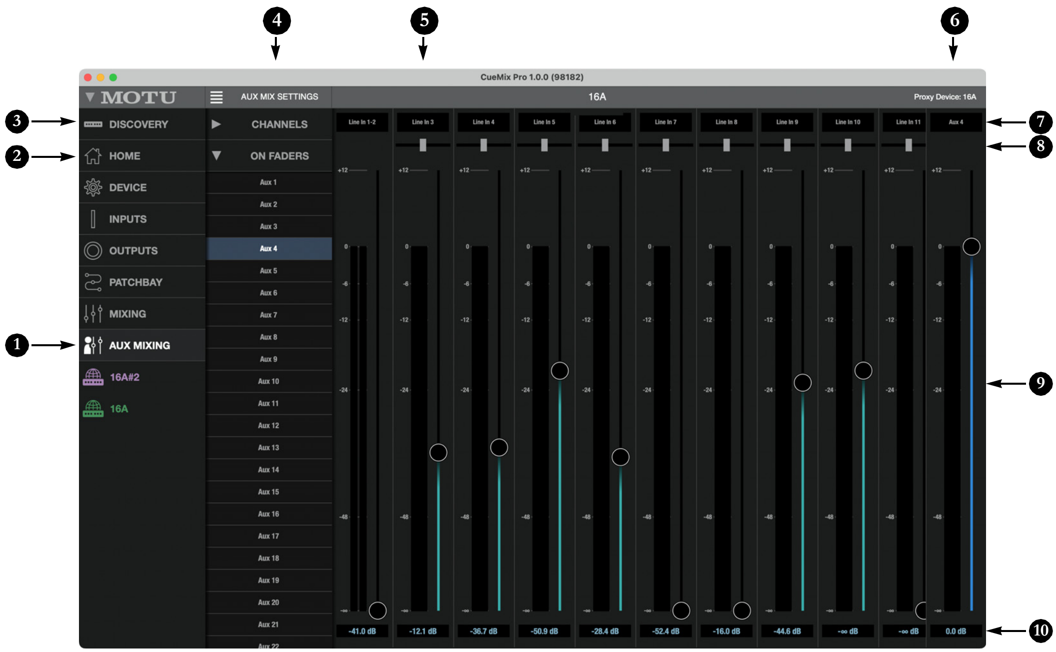
13. *Solo* the input or bus here. Soloed channels and buses are routed to the *Solo bus*, which can be monitored via the *Monitor 1-2* bus. See "Bus channel strips" on page 40. Glide horizontally across multiple channels to solo them in one gesture.

14. The Talkback section provides access to the Talkback controls here in the Mixing tab. See "Talkback" on page 48.

15. The *On Faders* section lets you temporarily convert the faders in the input channel strips into send faders for the chosen bus, providing a convenient way to route multiple inputs to the bus. This includes the reverb bus, for routing input signals to the 16A's built-in reverb. See "Reverb" on page 47. To bring the faders back to controlling input level, choose *Mains* in the On Faders list. When a bus is chosen in the On Faders list, the channel strip pinned to the right side of the window changes to the fader and channel strip for the chosen bus. Choose *Mains* to revert it back to the Main 1-2 bus.

16. The *Aux Buses* list provides controls for configuring aux buses as mono or stereo pairs. The *Reverb* bus is always stereo. You can also specify the sends for each bus as *pre-fader* (PRE on) or *post-fader* (PRE off). This setting applies globally to all send faders to the bus.

AUX MIXING TAB



1. The *Aux Mixing* tab provides quick access to the mixer's 26 aux buses and reverb bus, viewed one at a time. Choose a bus in the *On Faders* section of the settings sidebar (2) and then use the faders to directly mix the send levels for all 64 mixer inputs.

2. Use the *On Faders* list to choose which aux bus to work with here in the Aux Mixing tab. In this example, the faders for each input are sends to the *Aux 4* bus.

3. Use the *Channels* list in the sidebar to show or hide input channel strips here in the Aux Mixing tab.

4. Click the *Aux Mix Settings* menu icon to open (or close) this sidebar.

5. This is one of 64 *input channel strips*. It provides a send fader and pan control for the input. These two controls (volume and pan) are for the send only, not the input channel.

6. This is the bus fader for the aux bus currently being viewed here in the Aux Mixing tab (as determined by the On Faders selection). It is always pinned to the right side of the window.

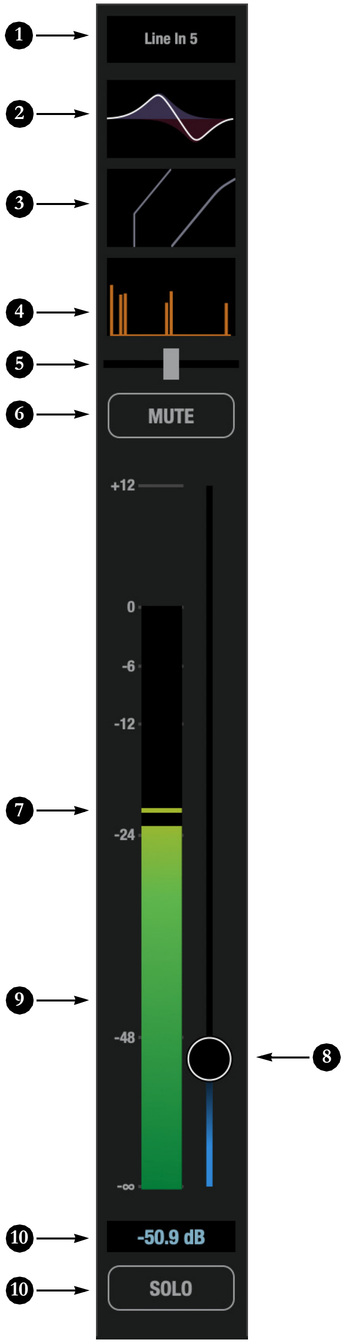
7. This is the source signal for the input.

8. This is the *Pan* control for mono input channels. Inputs that are grouped as stereo pairs do not have this pan control. To create stereo pairs, click the channel setting (item #5 on page 37) for one of the two channels, or any thumbnail.

9. Channel *send fader* and *level meter*. Double-click the fader to jump to unity gain or -∞. Level meters are pre-fader.

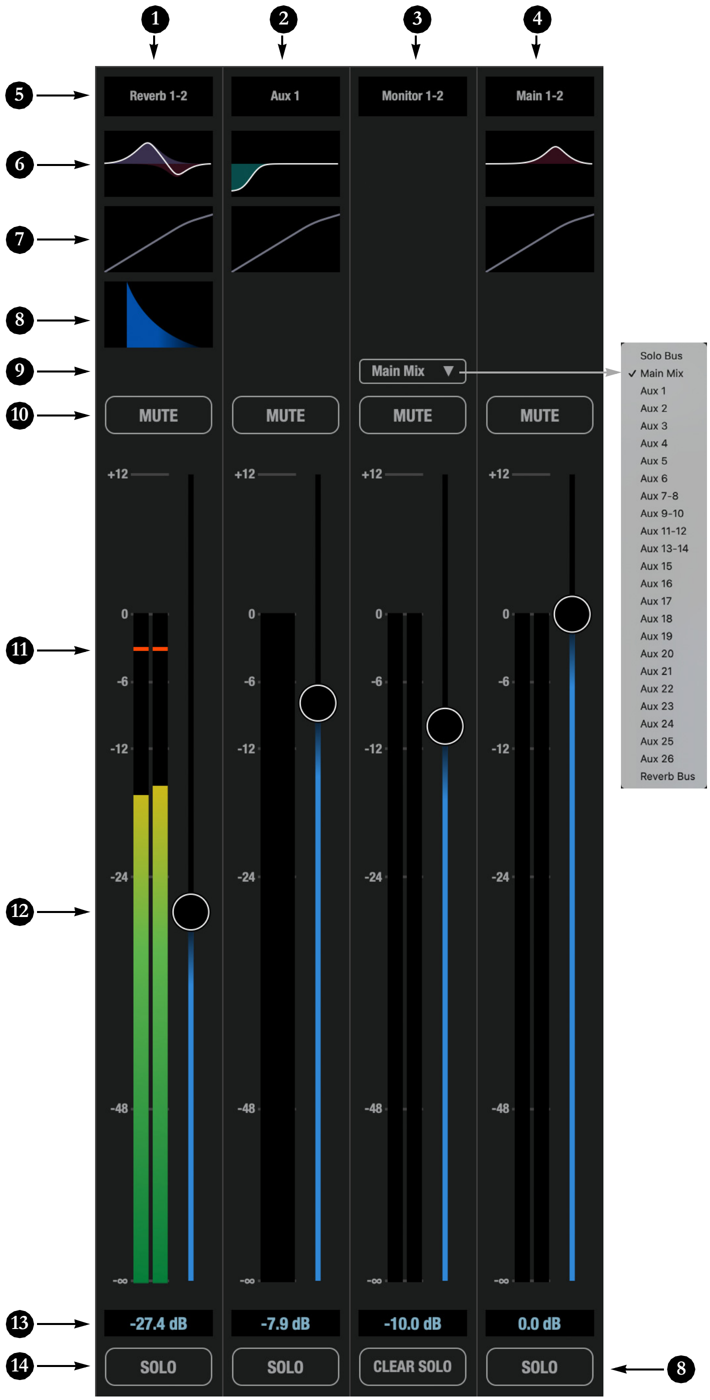
10. *Fader level* in dB.

INPUT CHANNEL STRIPS



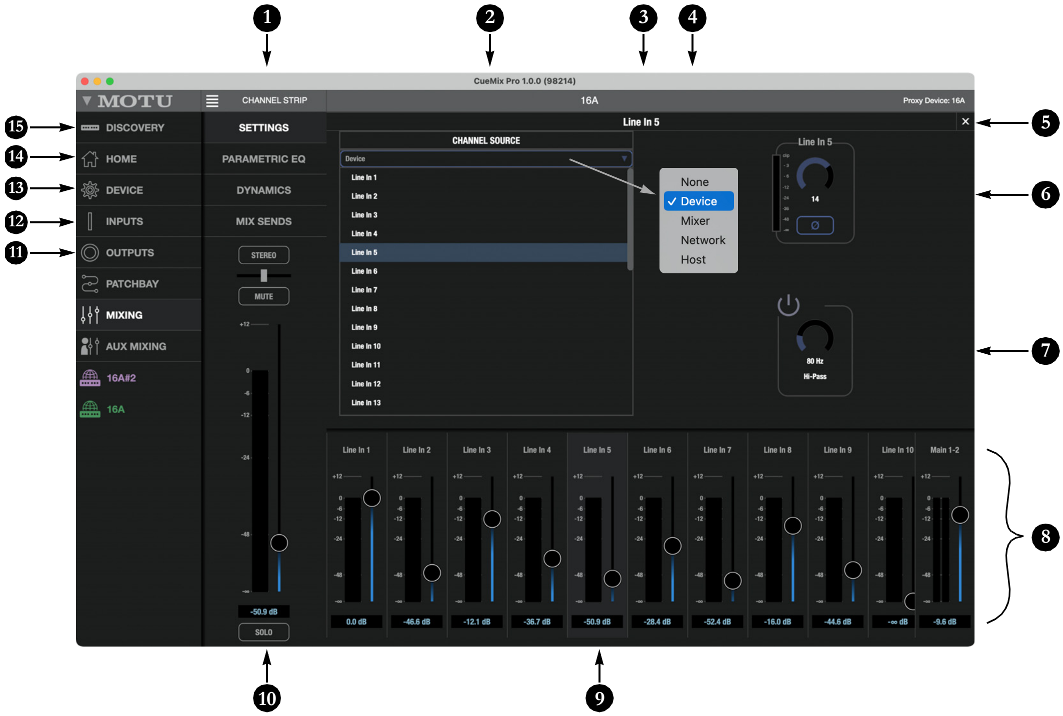
1. This displays the source signal for the input. Click it to access basic settings for the channel, such as the source signal and High Pass Filter (HPF). See "Input channel settings" on page 41.
2. Click this EQ thumbnail to access the parametric EQ and other channel settings. See "Multi-band parametric EQ" on page 45.
3. Click this dynamics processing thumbnail to access the *Gate*, *Compressor* and other channel settings. See "Compressor" on page 46 and "Gate" on page 46.
4. This is a *Send fader* thumbnail that shows any send faders that are up for the input. Click to access all send faders for the input, consolidated in one window. See "Mix Sends" on page 44.
5. *Pan control* for mono input channels. Inputs that are grouped as stereo pairs do not have this pan control. To create stereo pairs, click the channel settings (1) for one of the two channels or any thumbnail.
6. Channel mute. Glide horizontally across multiple channels to mute them in one gesture.
7. The peak/hold indicator shows where the signal has recently peaked.
8. Use the channel fader to control the input level. The fader range is from -∞ dB to +12 dB. Double-click the fader to jump to unity gain or -∞. Level meters are prefader.
9. The level meter for each input is pre-fader.
10. The fader value displayed numerically.
11. *Solo* the input here. Soloed channels are routed to the *Solo bus*, which can be monitored via the *Monitor 1-2* bus. See "Bus channel strips" on page 40. Glide horizontally across multiple channels to solo them in one gesture.

BUS CHANNEL STRIPS



1. The *Reverb* bus sends input signals to the reverb processor (8). The resulting stereo output from the reverb should be routed to a mixer input, where it can be added to the main mix or any aux mix (via sends).
2. The mixer has 26 *Aux* buses that you can use for flexible mixing and routing.
3. The *Monitor 1-2* bus is dedicated to the role of monitoring. You can use the menu (shown below) to make the Monitor bus follow the Main Mix, the Reverb bus or any Aux bus. When any input or bus is soloed, the Monitor bus temporarily follows the *Solo Bus* (until all solos are cleared). If you choose Solo Bus from this menu, the Monitor bus always follows the Solo Bus, even when nothing is soloed. Note that the Monitor bus has no dedicated EQ or dynamics processing, as those functions can be applied to the bus it is following (except the Solo bus, which would never need them).
4. This is the *Main Mix* bus for the mixer.
5. This is the *name* of the bus. Click it to access the settings for the bus.
6. Click this *EQ thumbnail* to access the parametric EQ and other bus settings. See “Multi-band parametric EQ” on page 45.
7. Click this *dynamics processing* thumbnail to access the bus compressor and other bus settings. See “Compressor” on page 46.
8. Click the *Reverb* thumbnail to access the settings for the reverb processor. See “Reverb” on page 47.
9. Use the *Follow* menu in the Monitor bus channel strip to choose which bus the Monitor bus should follow, as shown in the menu to the left. See item #3 above for further explanation of this menu.
10. Bus mute. Glide horizontally across multiple buses to mute them in one gesture.
11. -The peak/hold indicator shows where the signal has recently peaked.
12. Use the fader to control the bus level. Double-click to return to -∞ or unity gain.
13. The fader value displayed numerically.
14. *Solo* the bus here. Soloed buses are routed to the *Solo bus*, which can be monitored via the *Monitor 1-2* bus (9). Glide horizontally across multiple buses to solo them in one gesture.
15. Use the Clear Solo button in the Monitor bus channel strip to clear all solos.

INPUT CHANNEL SETTINGS



Click any of the items at the top of an input channel strip (items #1, 2, or 3 on page 39) to access the Channel Settings shown here.

1. Click the *Channel Strip* heading to show or hide this Sidebar, which displays basic settings for the channel and provides tabs for the channel's 4-band EQ, dynamics processing and send faders.

2. Choose the signal source for the input channel here. The menu at the top of the list lets you browse four possible banks of source signals as shown: *Device*, *Mixer*, *Network*, or *Host*. In fact, you can route signals to this mixer input in the Patchbay tab using those same banks of sources.

3. This is the mixer *Channel Name*.

4. Use the *channel source* bank menu to access the desired input signal for the mixer input.

5. Click this "x" to close the settings and return to the mixer.

6. These are the input settings for the selected source signal, if any. For example, if you choose a 16A analog input, you'll have access to the input gain and phase invert controls, as shown here. These are the same controls as available in the Inputs tab (item #2 on page 34).

7. *High-Pass Filter* (HPF) for the channel. The HPF is applied first, before any other processing.

8. The *Mixer pane* shows faders for all mixer inputs and busses, so you can access them while adjusting channel settings for a specific channel above. Click on any channel strip to access the settings for that input or bus.

9. This is the currently selected channel. Click any channel to view its channel strip settings.

10. This is the complete channel strip for the currently selected channel or bus. Use the tabs at the top of the strip to access basic settings, EQ, dynamics and bus sends.

11. The *Stereo* button links or unlinks the channel to an adjacent channel for stereo pairing.

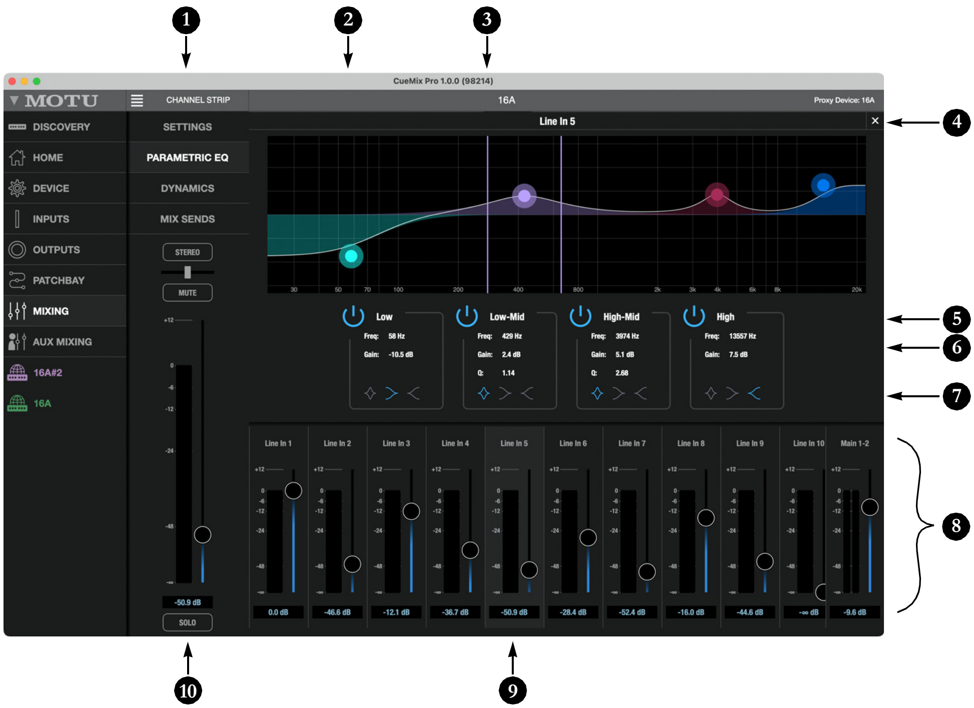
12. Click *Mix Sends* to access all *bus sends* for the channel.

13. Click *Dynamics* to access the channel's *Compressor* and *Gate*.

14. Click *Parametric EQ* to access the channel's *Parametric EQ*.

15. Click *Settings* to access the basic settings show above for the channel.

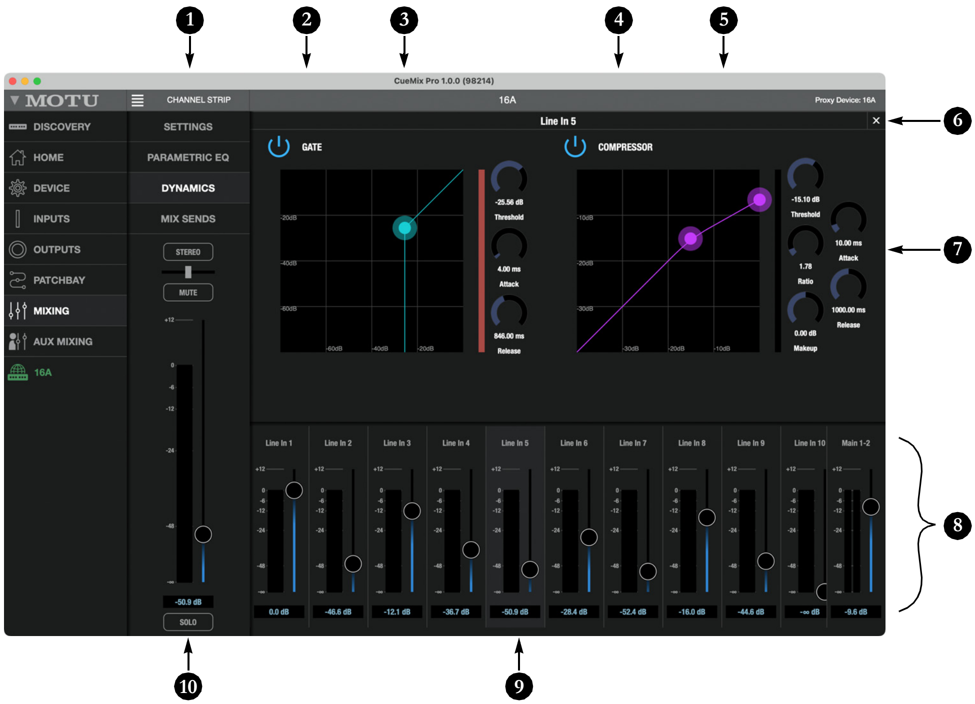
PARAMETRIC EQ



Click any of the items at the top of a channel strip (items #1, 2, or 3 on page 39 for an input or items 5, 6 or 7 on page 40 for a bus) to access the Parametric EQ and other settings for the channel.

1. Click the *Parametric EQ* tab here in the channel strip sidebar to access the 4-band parametric EQ settings shown here. For further information, see “Multi-band parametric EQ” on page 45.
2. Drag the *Frequency/Gain* handle for an EQ band to change the frequency and/or gain for the band.
3. Drag the *Q (bandwidth)* handles for an EQ band to change them. The handles are color-coded to match the color of their respective EQ band.
4. Click this “x” to close the settings and return to the mixer.
5. Click the “power” icon to enable or disable the EQ band.
6. Click values to edit them numerically. You can also drag vertically on them to change them.
7. Click the *Notch/Shelf* switches to toggle the filter type.
8. The *Mixer pane* shows faders for all mixer inputs and busses, so you can access them while adjusting the EQ for a specific channel above. Click on any channel strip to access the settings for that input or bus.
9. This is the currently selected channel. Click any channel to view its channel strip settings.
10. This is the complete channel strip for the currently selected channel or bus. Use the tabs at the top of the strip to access basic settings, EQ, dynamics and bus sends.

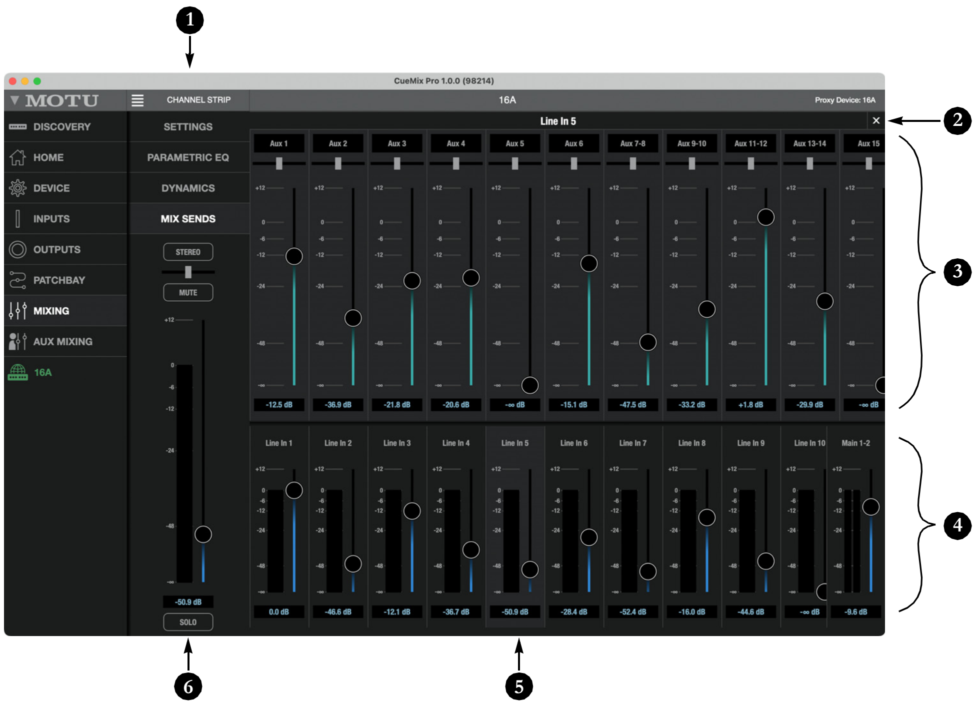
DYNAMICS PROCESSING



Click any of the items at the top of a channel strip (items #1, 2, or 3 on page 39 for an input or items 5, 6 or 7 on page 40 for a bus) to access the dynamics processing and other settings for the channel.

1. Click the *Dynamics* tab here in the channel strip sidebar to access the *Gate* and *Compressor* shown here.
2. The Gate processor is available on mixer input channels only. See "Gate" on page 46. Use the "power" button to enable or disable the gate.
3. Drag the *Threshold* handle to adjust it graphically.
4. The Compressor is available on all mixer inputs and busses. See "Compressor" on page 46. Use the "power" button to enable or disable the compressor.
5. Drag the *Threshold* and *Ratio* handles to adjust them graphically.
6. Click this "x" to close the settings and return to the mixer.
7. Click values to edit them numerically. You can also drag on them vertically to change them.
8. The *Mixer pane* shows faders for all mixer inputs and busses, so you can access them while adjusting the dynamics processors for a specific channel above. Click on any channel strip to access the settings for that input or bus.
9. This is the currently selected channel. Click any channel to view its channel strip settings.
10. This is the complete channel strip for the currently selected channel or bus. Use the tabs at the top of the strip to access basic settings, EQ, dynamics and bus sends.

MIX SENDS



Click any of the items at the top of an input channel strip (items #1, 2, or 3 on page 39) to access the Mix Sends shown here.

1. Click the *Mix Sends* tab here in the channel strip sidebar to access the send faders shown across the top of this window.
2. Click this "x" to close the settings and return to the mixer.
3. This scrolling pane shows pan sliders and send faders for each mix bus. To send input signal to the bus, just bring up the fader. Double-click the fader to jump to unity gain or -∞. Level meters are pre-fader.
4. The *Mixer pane* shows faders for all mixer inputs and busses, so you can access them while adjusting the sends for a specific channel above. Click on any input channel strip to access the settings for that input. Sends are not available for buses. (You can't send a bus to another bus).
5. This is the currently selected channel. Click any channel to view its channel strip settings.
6. This is the complete channel strip for the currently selected channel or bus. Use the tabs at the top of the strip to access basic settings, EQ, dynamics and bus sends.

MIXER EFFECTS

The high-pass filter, parametric EQ, compressor, gate, and reverb processor in the 16A are available when operating the unit at 1x and 2x samples rates. At 4x sample rates (176.4 or 192 kHz), all effects processing is disabled, but the mixer still provides 64 inputs and 32 buses.

DSP-driven mixing and effects

The 16A effects are driven by a powerful DSP that delivers 32-bit floating point precision and plenty of bandwidth for no-latency processing. Effects can be applied when operating as an audio interface or as a stand-alone mixer.

Advantages over host-based mixing and processing

The hardware mixer in the 16A provides several advantages over mixing and processing in your host audio software:

- No buffer latency. The DSP-mixer provides the same near-zero latency throughput performance as a conventional digital mixer. Effects processing doesn't impact your computer's CPU.
- DSP mixing and routing can be maintained independently of individual software applications or projects.
- Monitoring is maintained during computer related disruptions (such as switching from one DAW project to another, for example).
- DSP-driven mixing can function without the computer, allowing the 16A to operate as a portable, stand-alone mixer with effects.

Accessing mixer effects

To access the EQ, compressor or gate for a channel, click its thumbnail at the top of the channel strip (item #2 or 3 on page 39).

MULTI-BAND PARAMETRIC EQ

All input channels and buses provide four bands of center-frequency parametric EQ filtering as shown on page 42. All bands include shelf filtering options.

Enabling EQ bands

Each band has an *enable/disable* button (item #5 on page 42), allowing you to enable as few or as many bands as needed.

EQ filter controls

The EQ filters have three controls (item #6 on page 42):

Control	unit	range
Frequency	Hertz	20 to 20,000
Gain	dB	-20.00 to +20.00
Q (bandwidth)	n/a	0.45 to 10.0

Click a value to edit it numerically. Or drag on it vertically to change it. Or drag its corresponding control handle in the graph above. Controls for each EQ band are color-coded.

EQ filter characteristics

EQ is one of the most widely used processing tools and can be applied to many different situations, from minor corrective tasks to creative tone sculpting. The multi-band EQ has been designed to be flexible enough to cover a broad range of applications. By adjusting Gain and Bandwidth together, you can emulate the smooth and musical character of classic analog EQ circuits.

Filter types

The shelf filter options are similar to those found in most conventional parametric EQs and can be used to filter out low or high frequencies.

GATE

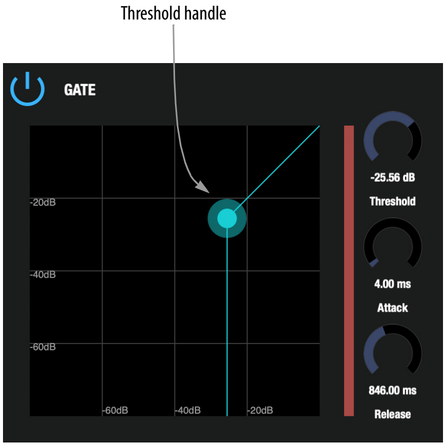


Figure 6-1: The Gate.

To access the gate, click the compressor/gate thumbnail (item #3 on page 39).

The gate silences the signal when the input signal's level drops below the *Threshold*.

The rate at which the gate responds, (opens to let signal through) is determined by the *Attack* parameter. With a short Attack time, the gate will open as soon as the signal crosses the Threshold; with longer Attack times, the gate will gradually open, much like a fade-in.

When the input level falls back below the Threshold, the time it takes for the gate to close (how quickly the signal is attenuated), is determined by the *Release* parameter. Short Release times will close the gate quickly, abruptly attenuating your signal, versus longer release times, which will gradually attenuate your signal, like a natural fade-out.

COMPRESSOR

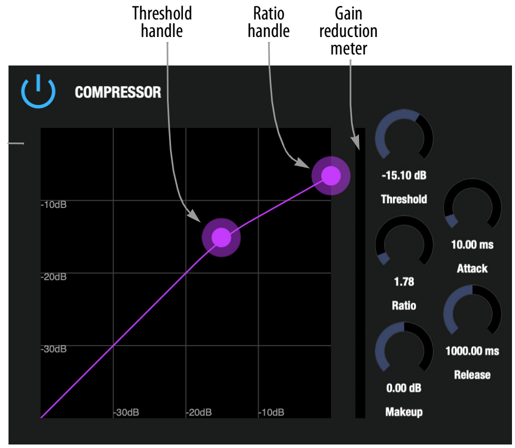


Figure 6-2: The Compressor.

To access the compressor, click the compressor/ gate thumbnail (item #3 on page 39).

The *Compressor* lowers the level of the input when amplitude of the signal is above the Threshold. The amount of attenuation is determined by the *Ratio* and the input level. For example, if the input is 6 dB above the *Threshold* and the Ratio is 3:1, the compressor will attenuate the signal to 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", a term used to describe the sound of the compressor when the average input level quickly fluctuates above and below the Threshold. These issues can be addressed by adjusting the parameters. The *Gain reduction meter* (Figure 6-2) displays the attenuation applied by the compressor.

REVERB

To access the reverb settings: scroll to the Reverb bus (item #1 on page 40) and then click the reverb thumbnail, as shown below.

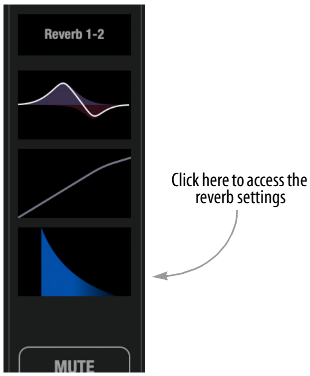


Figure 6-3: Accessing the Reverb processor.

Reverb settings

The Reverb processor (Figure 6-4) provides *Small*, *Medium* and *Large* room sizes, along with *Pre-Delay* (see below), *Damping*, *Decay* (length) and *Width* (stereo image) settings.

Predelay

Predelay is the amount of time before the acoustic energy from the source returns to the listener, after reflecting off the surfaces of the listening space. The very first reflections helps you perceive information about the listening space, (size, distance, surface type, etc.) In large rooms, it takes a while (on the order of milliseconds) before the first reflections return to the listener. Predelay is useful for adding clarity, as it delays these reflections, before the onset of full reverberation. For example, with pre-delay added to vocals, the reflections won't start until after the initial sound of a word has been sung.

Routing inputs to the reverb processor

The reverb processor is a single, independent unit that provides stereo reverb. Use sends to route input signals to it ("Aux Mixing tab" on page 38 and "Mix Sends" on page 44). All incoming signals to the reverb processor are mixed and processed together. The resulting stereo output from the reverb should be routed to a mixer input, where it can be added to the main mix or any aux mix (via sends).

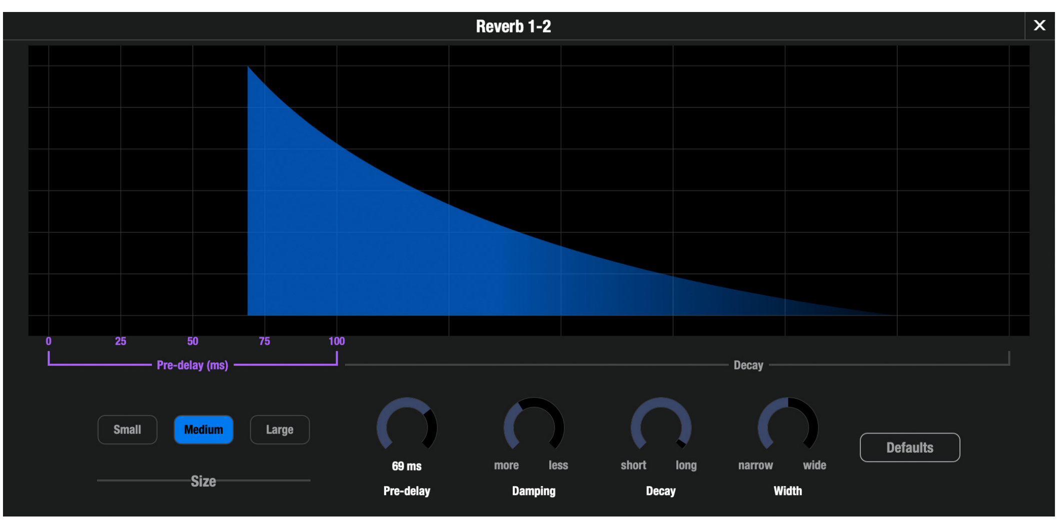


Figure 6-4: The reverb processor.

A/B/C MONITORING

The CueMix Pro Outputs tab (page 35) provides controls for primary, secondary, and auxiliary monitors in your studio, labeled A, B, and C, respectively.

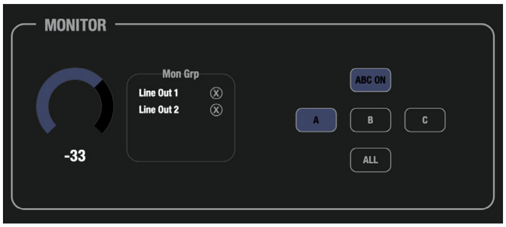


Figure 6-5: A/B/C monitoring.

Output connections for A/B/C monitoring

Connect your speakers as follows:

- Monitor pair A --> Line Outs 1-2
- Monitor pair B --> Line Outs 3-4
- Monitor pair C --> Line Outs 5-6

Enabling A/B/C monitor mode

To enable or disable A/B/C monitor mode, press the *ABC ON* button (Figure 6-5). When enabled, the output channels in monitor group A, B, and C share the same audio signal (assigned to the channels in group A).

Monitor A/B/C select

To select a monitor pair and mute the other pairs, press A, B or C (Figure 6-5). Press the *ALL* button to hear all three sets of monitors simultaneously.

A/B/C volume control and separate trim

Control the volume of all three monitor pairs with the large volume knob (to the right of the LCD) on the front panel of the 16A. This knob controls all three monitor pairs. To adjust their volume relative to one another, use their trim controls in the Output tab (item #7 on page 35).

TALKBACK

Talkback allows an engineer in the control room to temporarily dim or mute all audio and talk to musicians during a recording session. Talkback requires a microphone located in the control room, near the engineer.

Figure 6-6: talkback settings can be found in the CueMix Pro Home tab.

Talkback setup

To set up talkback:

1 Connect a microphone to a preamp and then connect the output of the preamp to any 16A analog input.

2 In CueMix Pro's Home tab, choose the mic input from the Source menu.

3 In the Destinations menu, check the outputs you wish to send the talkback signal to.

4 Adjust the Talkback settings as explained below.

Talkback settings

Talkback has the following settings (Figure 6-6).

Talk

Press and hold the *Talk* button (Figure 6-6) to engage the talkback mic.

Latch

When *Latch* is engaged (Figure 6-6), the Talk button remains engaged when you click it, until you click it again to disengage, so you don't have to hold it down while speaking.

Level

The Talkback *Level* setting controls the volume of the talkback signal, as an additional gain stage after the talkback mic preamp gain (and pad, if any).

Dim

If you are feeding a monitor mix to the musicians on the same Aux bus as your talkback mic, use the *Dim* knob (Figure 6-6) to control how much the monitor mix will be attenuated when talkback is engaged. This gives you control over the relative volume between the talkback mic signal and all other audio on the mix bus. To control overall volume of everything, use the bus fader.

Mic effects

Click the *Mic effects* thumbnail (Figure 6-6) for quick access to the EQ, compressor and gate effects for the analog input you are using for talkback. These are the same settings found in the mix tabs for that input (items #2 and 3 on page 39).

WORKING WITH OTHER MOTU AVB INTERFACES

If earlier-generation MOTU AVB interfaces are present on the network, they appear in the sidebar. Click on them to access basic settings in the Device tab. Unavailable tabs will be grayed out.

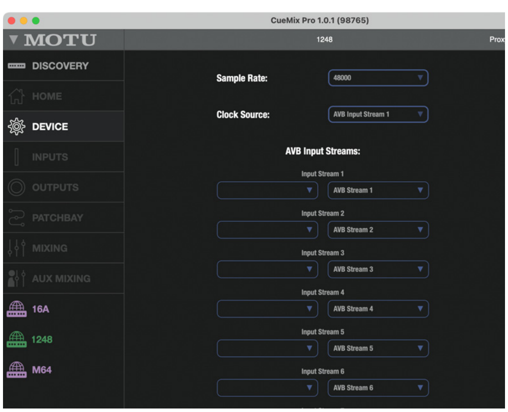


Figure 6-7: A MOTU 1248 selected in the sidebar. Basic network-related settings are available in the Device tab.

FIRMWARE UPDATER

From time to time, MOTU may release firmware updates for the 16A to improve operation and add new features. To check to see if there is a firmware update available, choose *File menu > Open Firmware Updater* to launch the updater app.

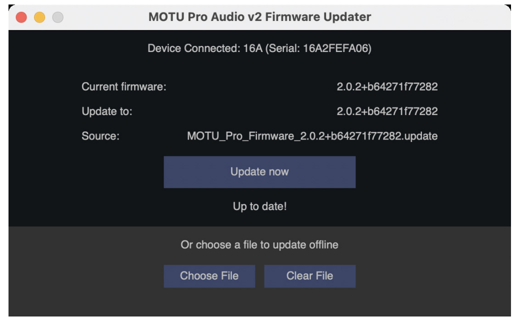


Figure 6-8: The 16A Firmware Updater app.

If your computer has access to the internet, the updater app will check to see if an update is available and will notify you, if so. Or, you can download a firmware image from motu.com/download and then load the firmware image with the *Choose File* button.

Once the firmware image is loaded and ready to go, click the *Update Now* button (Figure 6-8).

The 16A unit will reboot and the screen will display *Update...* A progress bar in the Firmware Updater app indicates the time remaining. When complete, the 16A unit will restart on its own and is now ready for normal operation.

☛ Previous settings and presets are preserved during the firmware update process, so you don't need to worry about saving and restoring settings.

THE CUEMIX PRO IOS APP

CueMix Pro is also available as an app for iOS and allows you to control the 16A from your iOS device. Simply download it from the App Store. Also see "iOS setup (USB-C)" on page 19 and "iOS setup (Lightning)" on page 19.

CHAPTER 7 Working with Host Audio Software

The 16A provides multi-channel audio input and output for Core Audio compatible audio applications on the Mac and ASIO or Wave compatible applications on Windows, including MOTU's Digital Performer and Performer Lite, Apple's Logic Pro and GarageBand, and other third-party software applications such as Ableton Live, Avid Pro Tools, Cockos Reaper, Propellerhead Reason, Steinberg Cubase and Nuendo, PreSonus Studio One, Bitwig, and others.

Performer Lite is available as a free download for 16A owners from their motu.com account page. For complete information about all of Performer Lite's powerful workstation features, refer to the *Performer Lite User Guide.pdf* found in the Help menu of the Performer Lite application.

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

PREPARATION

Install your host audio software first if you haven't already done so, and complete these chapters before proceeding:

- chapter 3, "Software Installation" (page 15)
- chapter 4, "Hardware Installation" (page 18)

RUN THE CUEMIX PRO APP

Before you run your host audio software, launch CueMix Pro to configure your MOTU hardware.

Sample Rate

Choose the desired sample rate for the 16A (item #4 in the Devices tab on page 33) and your host audio software. Make sure the sample rates for the hardware and software match. Newly recorded audio will have this sample rate.

☛ Optical channels are disabled when the interface is operating at a 176.4 or 192 kHz. In addition, channel counts for host computer I/O and networking are reduced.

Clock Mode

The *Clock Source* setting (item #5 in the Devices tab on page 33) is important because it determines the master digital audio clock for your system.

If you do not have any digital audio connections to your MOTU device (you are using the analog inputs and outputs only), and you will not be resolving your host software to optical or another external clock source, choose *Internal*.

If you have devices connected to the optical ports, see "Choosing a clock source for optical connections" on page 22.

If you are resolving your system to word clock, see "Syncing word clock devices" on page 23.

CHOOSE THE 16A

Once you've made the preparations described so far in this chapter, you're ready to run your host audio software and choose the 16A as your audio interface ("soundcard").

For macOS audio software

For audio software running under macOS, go to the menu item or preference where you choose the audio device (Core Audio driver) you wish to use, and then select the 16A by name.

For Windows audio software

For audio software running under Windows, go to the menu item or preference where you choose the ASIO driver you wish to use, and then choose *MOTU 16A*. If your host audio software doesn't support ASIO, choose the *16A* Wave driver instead.

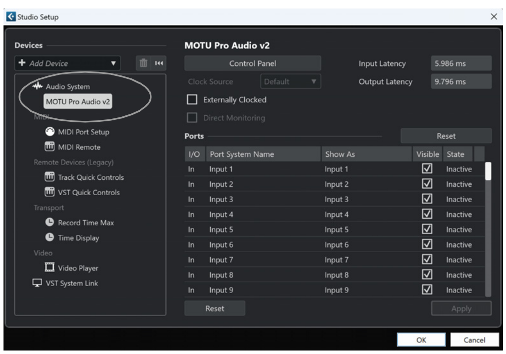


Figure 7-1: Choosing the MOTU Pro Audio ASIO driver in Cubase.

Where to go in popular audio hosts

Here is the location for this setting in various popular audio software host applications:

Host software	Location for choosing the 16A
Digital Performer and Performer Lite	Setup menu > Configure Audio System > Configure Hardware Driver
Pro Tools	Setup menu > Playback Engine or Current Engine
Logic Pro	Preferences > Audio tab > Devices tab > Core Audio tab
Garage Band	Garage Band menu > Preferences > Audio/ MIDI > Audio Output/Input menus
Cubase and Nuendo	Device Setup > Devices list > VST Audio System menu
Live	Preferences > Audio tab
Reason	Preferences > Audio preferences
Reaper	Preferences > Audio prefs > Devices
Studio One Pro	Preferences > Audio Setup

Other audio software

Consult your software's manual for further information.

CHANNEL COUNTS

Available channels to and from the host computer are as follows, based on sample rate and driver versus USB class compliant operation::

Sample Rate (kHz)	44.1 / 48	88.2 / 96	176.4 / 192
Thunderbolt/USB4	128	128	64
USB3 (driver)	128	64	32
USB3 (class)	128	64	32
USB2 (driver)	64	32	18
USB2 (class)	64	32	18

REDUCING MONITORING LATENCY

Monitoring latency is a slight delay caused by running an input signal through your host audio software and back out. 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 input on the 16A, passes through the interface hardware into the computer, through your host audio software, and then back out to an output.

Monitoring through the 16A

If you don't need to process a live input with plug-ins, the easiest way to avoid monitoring latency is to disable your DAW's live monitoring feature and instead use the digital mixer in the 16A to route the input directly to your outputs. For details, see "Mixing tab" on page 37. The mixer in the 16A even provides zero latency effects processing (EQ, compression and reverb), which can be applied to the signal.

Direct hardware playthrough / Direct ASIO monitoring

When managing your live monitor mix through the 16A mixer, remember to disable your DAW's live monitoring features, so that you won't hear record-enabled tracks in your DAW. Also note that the 16A does not support *Direct Hardware Playthrough* in Digital Performer, or the *Direct ASIO Monitoring* feature (or similar) offered and other DAWs, which lets you control no-latency hardware monitoring from within the host application. Instead, you can use the CueMix Pro app mixer ("Mixing tab" on page 37) to set up these monitor mixes manually.

If you don't require any effects processing on the input signal (no reverb or compression, for example), all this takes is one click on a fader to route the input being recorded to the output you are using for monitoring.

If you are recording a mono input that you'd like to monitor in stereo, or if you need to apply effects to the monitored signal, you can use the 16A mixer for that, too. Use the mix tabs and reverb mix (page 37) to apply effects as desired, and perhaps include other channels to the mix.

Monitoring through your host audio software

If you *do* need to process a live input with host software plug-ins, or if you are playing virtual instruments live through your MOTU audio hardware, you can significantly reduce latency by adjusting the audio buffer setting in your host audio software, as explained in the next section.

☛ It is important to note that monitoring delay has no effect on the recording, or playback, of audio data from disk. The actual recording and playback is extremely precise, it is only the monitoring of your live input signal which may be delayed.

Adjusting your host software audio buffer

Buffers are small bundles of audio data. The 16A "speaks" to your computer in buffers, rather than one sample at a time. The size of these buffers determine how much delay you hear when monitoring live inputs through your audio software: larger buffers produce more delay; smaller buffers produce less.

Adjusting buffer size on macOS

Under macOS, audio I/O buffer size is handled by the host audio application (not by the 16A 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. Here are a few examples.

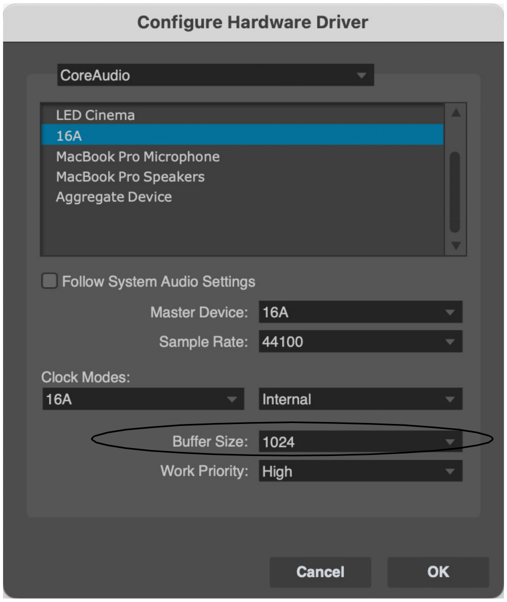


Figure 7-2: In Digital Performer and Performer Lite, choose Setup menu> Configure Audio System> Configure Hardware Driver to open the dialog shown above and access the Buffer Size setting.

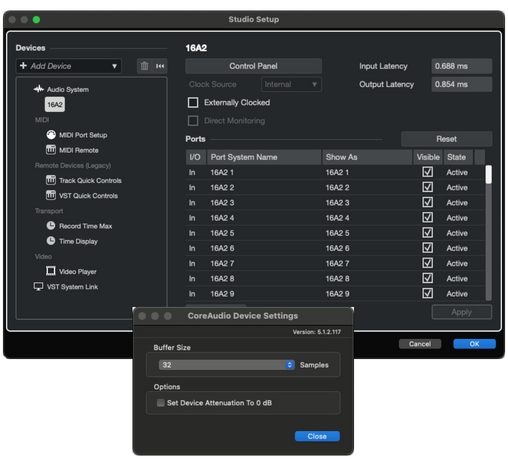


Figure 7-3: In Cubase or Nuendo, choose Devices menu > Device Setup. Select your interface (16A), then click the Control Panel button to access the window above and the Buffer Size setting.

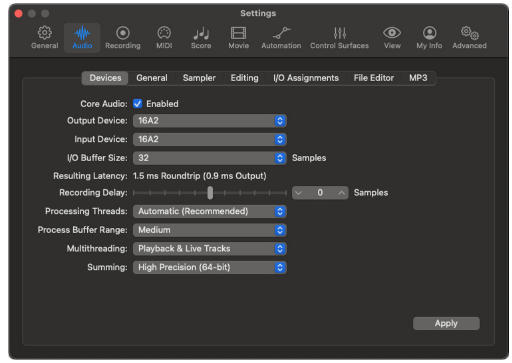


Figure 7-4: In Logic Pro, go to the Audio Driver preferences to access the Buffer Size option shown above.

Adjusting buffer size on Windows

On Windows, the buffer size is adjusted in the CueMix Pro app Device tab (items #7 and 8 in the Devices tab on page 33).

Lower latency versus higher CPU overhead

Buffer size has a large impact on the following:

- Monitoring latency
- The load on your computer's CPU
- Responsiveness of transport controls and effect knobs in Performer Lite, Digital Performer or other audio software.
- Real-time virtual instrument latency.

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 monitoring 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.

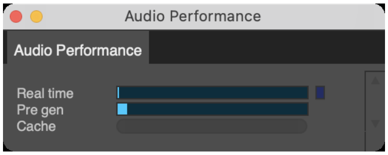


Figure 7-5: When adjusting the buffer size to reduce monitoring latency, watch the 'Real time' meter in Digital Performer's Performance Monitor. If you hear distortion, or if the meter is peaking, try raising the buffer size.

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.

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.

LOOPBACK

Loopback is the process of sending audio output from your computer back to the computer so that you can capture the signal in your host software, or stream it live to the web.

Direct loopback in the Patchbay

The simplest way to set up loopback is to route host computer source channels to host computer destination channels directly in the Patchbay, as shown in Figure 7-6. This method works fine if you don't need to include other source signal in the loopback channels. Note that you can set up as many loopback channels as needed: you're not limited to just two.

Using an aux bus for loopback

If you use an aux bus for loopback, you can include additional channels in the loopback mix:

1 Choose an aux bus for loopback and make it stereo. See "Bus channel strips" on page 40 and "Input channel settings" on page 41.

2 In the Patchbay tab, route any computer output channels you wish to include in the loopback signal to mixer input channels. These computer channels can include audio tracks, virtual instruments, or any other audio output from your audio software.

Figure 7-6: Creating loopback routing in the Patchbay. In this example, Host output channels 63-64 are being routed to Host input channels 63-64.

3 Make sure the input channel faders are up enough to allow plenty of signal.

4 Using sends, route each computer input channel to your loopback aux bus.

5 To include other mixer channels in the Loopback mix, such as live inputs from microphones or instruments connected to the 16A, do the same for them as well.

6 In the Patchbay tab, route the output of the Loopback bus to any desired computer input channel pair.

The Loopback mix is now routed back to the computer, where it can be recorded, monitored, streamed, etc.

☛ Some podcasting software applications on the market can only see the first 1-2 or 1-4 channels from an audio interface. In this case, the Loopback aux bus will need to be assigned to computer input channels 1-2 or 3-4.

☛ Remember, be careful! When monitoring loopback channels and live inputs, your host software can cause loud feedback loops. Be sure to disable the monitoring of loopback tracks to avoid feedback.

WORKING WITH MULTIPLE INTERFACES

If you have multiple 16A interfaces (or other AVB equipped interfaces), the best way to use them with your host software is to connect them via AVB to the 16A unit connected to your computer, as explained in "Connecting multiple 16A's to a host" on page 20. The AVB interfaces can then stream audio to and from the host interface, which sends the streams to and from the host computer. This approach provides a great deal of flexibility in how you can physically connect the interfaces, and it provides high channel counts with low latency. See "Setting up the 16A for networking" on page 61 for further information.

CHAPTER 8 Optical Expander Presets

OVERVIEW

The 16A has the ability to function as an optical expansion interface. You can connect it to other optical-equipped devices to route the 16A's analog inputs and outputs to and from the other device. Up to 16 channels are supported at 1x sample rates (44.1 or 48 kHz) and 8 channels at 2x rates (88.2 or 96 kHz).

☛ The Optical Expander factory presets are designed to support the 8-channel ADAT format for both Bank A and B. If you would like to use stereo TOSLink for Bank A input or output, you will need to change the Bank A format manually after choosing an Optical Expander preset, as well as any necessary changes in the Patchbay routing.

MAKING CONNECTIONS

To route the 16A's analog inputs to the other device, connect the 16A Optical OUT A port to the other device's Optical IN port. If the other device has a 2nd bank of optical, connect the 16A's Optical OUT B to the other device's 2nd Optical IN bank.

To route audio from the other device to the 16A's analog outputs, connect the other device's Optical OUT to the 16A's Optical IN A port. Similarly, if the other device as a 2nd bank of optical, connect it's 2nd Optical OUT bank to the 16A's Optical IN B port.

CHOOSING AN OPTICAL EXPANDER PRESET

Before you choose an optical expander preset, you may wish to save the current state of hardware, so you can return to it later. To do so, turn the MENU knob on the front panel and choose *Presets*. Then use the SELECT button and MENU knob to save to an empty preset slot. Use the BACK button to return to the meter screen when you are finished.

To choose an optical expander preset, turn the MENU knob and choose *Optical Expander Presets*. Then use the SELECT button and MENU knob to select the optical expander preset that matches the sample rate you wish to work at:

- Optical Expander @ 44.1k
- Optical Expander @ 48k
- Optical Expander @ 88.2k
- Optical Expander @ 96k

Optical expander preset settings

The *Optical Expander* presets leave most of the 16A settings unaffected, including basic device settings, mixing, effects, etc. They do affect settings that are directly relevant to optical expansion, as follows:

- All analog inputs are patched directly to both optical output banks.
- Both optical input banks are patched directly to analog outputs.
- All analog input gain and output trim settings are zeroed out (no gain or trim).
- All analog outputs are assigned to the Monitor Group, such that all analog output can be controlled with the main volume knob.

SETTING THE SAMPLE RATE

In the CueMix Pro Device tab, be sure that the sample rate setting (item #4 on page 33) for the 16A matches both the preset you chose and the sample rate of the other device. At the 1x sample rates (44.1 and 48 kHz), each optical bank provides 8 channels. At the 2x rates (88.2 and 96 kHz), each bank provides 4 channels.

SETTING THE CLOCK SOURCE

When operating as an Optical Expander, the 16A transfers digital audio channels over it's optical connections. It is therefore crucial to make sure that the clocks in the two devices are properly resolved. See the following sections for further information:

- "Synchronization" on page 22
- "Syncing optical devices" on page 23

CHANNEL MAPPING AT 44.1 OR 48 KHZ

At 1x sample rates, the 16A's two optical banks are mapped as follows:

Optical Input	Destination
Bank A Inputs 1-8	Analog Outputs 1-8
Bank B Inputs 1-8	Analog Outputs 9-16

Source	Optical Output
Analog Inputs 1-8	Bank A Outputs 1-8
Analog Inputs 9-16	Bank B Outputs 1-8

CHANNEL MAPPING AT 88.2 OR 96 KHZ

At 2x sample rates, the 16A's two optical banks are mapped as follows:

Optical Input	Destination
Bank A Inputs 1-4	Analog Outputs 1-4
Bank B Inputs 1-4	Analog Outputs 5-8

Source	Optical Output
Analog Inputs 1-4	Bank A Outputs 1-4
Analog Inputs 5-8	Bank B Outputs 1-4

CHAPTER 9 Networking

OVERVIEW

The Audio Video Bridging (AVB) network ports on your MOTU interface open up a world of possibilities for creating expanded, customized audio network systems.

ABOUT AVB

Audio Video Bridging (AVB) is an extension of the Ethernet networking standard developed by the IEEE (1722.1 standards committee) specifically engineered for high-performance audio and video networking.

☛ You may also hear AVB referred to as *AVB/TSN* or simply *TSN* because AVB is based on a set of timing protocols referred to as *Time Sensitive Networking*.

The pro audio industry has seen many network-related protocols come and go over the years, but AVB has emerged as a widely-adopted open standard that brings together the worlds of networking technology and high-end audio. Here is a brief summary of some of the immediate benefits of AVB for you, as a MOTU AVB interface user:

- **An open industry standard** — AVB has been developed by the IEEE as an international standard specification, like AES/EBU and other similar IEEE-driven protocols. It is not proprietary or controlled by one company.

- **High channel counts** — AVB provides hundreds of network channels.

- **Extremely low, predictable latency** — AVB guarantees very low-latency real-time performance.

- **Guaranteed Quality of Service** — AVB's Stream Reservation Protocol provides guaranteed Quality of Service (QoS) for each and every audio stream. If the network cannot continuously maintain every bit of every sample in the audio stream, it will not allow you to make the network connection in the first place. AVB streams are prioritized over other network traffic to ensure high performance.

- **Network-wide clocking and sync** — AVB devices all clock together over your network for better-than-sample-accurate phase lock across all connected devices. Timing accuracy is down to the nanosecond.

- **Plug-and-play operation** — AVB has been designed from the ground up to provide automatic device discovery, enumeration, and connection management. Just plug your MOTU AVB interfaces into a standard AVB switch and go. If you wish to make stream connections and have the ability to select media clock, you must use the CueMix Pro app, or some other AVB controller. You don't need an IT professional to configure the network. AVB is a self-managing network protocol.

- **Shared operation with standard Ethernet** — AVB cooperates with standard Ethernet traffic on the network, so you can use traditional Ethernet devices like wireless routers, switches, or any other non-AVB-aware device on the same AVB network as your AVB devices.

- **Support for existing network infrastructure** — Replace your existing switches with standard AVB-compatible switches, and your CAT-5e or CAT-6 wired infrastructure now supports AVB.

- **Long cable runs** — A single AVB network connection can run up to 100 meters with a standard copper wire CAT-5e or CAT-6 cable. Fiber-optic cable runs can be much longer. With daisy-chains and multiple switches, you can create a network that covers very large distances, if necessary. You can connect up to seven "hops" (device-to-device connections).

MOTU'S AVB IMPLEMENTATION

MOTU engineering has faithfully implemented the IEEE AVB standard for MOTU AVB products. This means that MOTU devices are inter operable with compliant 3rd party AVB-compatible devices. This includes current IEEE 1722.1 compatible devices, earlier-generation MOTU AVB devices, and 3rd-party devices that support the legacy AM8-24 Stream Format.

In addition, MOTU has fine-tuned AVB operation among MOTU AVB devices for optimum performance, within the AVB specification. Here is a brief summary of advantages you will enjoy when using MOTU AVB devices together in a network:

- **Up to 256 channels of host I/O** — MOTU AVB interfaces (depending on the model) can support up to 256 simultaneous channels of audio I/O (128 in, 128 out) to and from the entire network through Thunderbolt or USB3.

- **Support for multiple computer hosts** — All computers and all network devices run in sync with each other, resolved to the network's master clock.

- **Gigabit Ethernet** — The MOTU AVB Switch delivers 1 Gbit Ethernet performance, which provides substantially higher bandwidth than 100 Mbit Ethernet. This allows you to have many more devices on the AVB network.

- **Over 600 channels of network audio** — A gigabit AVB network can stream over 600 channels of audio throughout the network (at 1x sample rates). The 16A can simultaneously broadcast and listen to 128 audio channels across 32 AVB streams.

- **Very low network latency** — Network latency is the time it takes for a data packet (such as an audio sample) to be transmitted over the network. Standard AVB network latency is 2 milliseconds (ms).

- **Flexible network topology** — MOTU AVB supports both daisy-chaining and hierarchical tree structures for maximum flexibility.

- **Cross-platform app** — MOTU AVB devices can be controlled from the CueMix Pro app, which runs on macOS, Windows and iOS.

- **MOTU AVB Switch**— The affordable six-port MOTU AVB Switch provides economical AVB networking for a wide range of small to medium sized AVB networks.

NETWORKING EXAMPLES

Here are just a few examples of what is possible.

Personal studio expansion

Let's say you have a 16A mounted in a rack next to your computer. You could add an 8M interface and position it across the room, near your drum kit, for placing up to 8 mics on the drums. All the mic cabling is kept near the drums, and you have one simple, clean network cable running back to your computer system. Despite the distance, the two interfaces operate as a seamless system, controlled from your computer or iPad.

Studio installation
Networking is ideal for studio installation because you can position interfaces at strategic locations. Running cables becomes much simpler and more cost effective. Not only does a setup like this give you access to all I/O from your computer, even multiple computers, you can also route audio from any input to any output across devices with near zero latency. You can also route audio from one computer to another with very low latency. As a simple example, you could deploy several interfaces across multiple rooms in a recording studio.

Large studio facility
In a larger studio facility, you could build audio network neighborhoods similar to the studio installation described earlier in multiple rooms, even multiple floors, with multiple computers and Wi-Fi control from anywhere in the facility. All computers and devices can see each other and you can stream audio anywhere on the network with near-zero latency, as if any two devices were connected directly to each other.

Concert systems
Concert systems must be flexible so they can adapt to each new venue while on tour. Because of its modular nature, AVB networking allows you to design systems that are scalable and easy to adapt to each venue. You can easily bring devices on and off line, rerouting audio stems as needed.

By connecting multiple computers to the network, you can set up backup playback systems on the network. For example, in a concert setting, if one computer system goes down, the backup system can be brought on line instantly through the same network infrastructure.

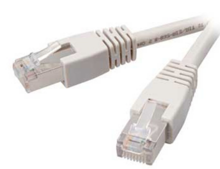
Traditionally, live performance setups often have separate domains for front of house mixing, monitor mixing, computer backline, and other systems. With MOTU AVB networking, these systems can be unified on the same network, opening up many possibilities for shared resources and mixing/routing responsibilities, especially from multiple sources (laptops, iPads, tablets, etc.) MOTU AVB's very low latency makes it particularly suitable for line arrays and sound reinforcement.

Large-scale venues
With long cable runs and industry standard networking infrastructure, MOTU AVB systems are well-suited for large-scale commercial installations such as arenas, stadiums, theme parks, clubs, casinos, houses of worship, broadcast facilities, schools, universities, and so on. Audio streams can travel long distances with sub-millisecond latency through as many as seven switches. Audio can be distributed from a centralized location to anywhere in the venue.

A QUICK GUIDE TO NETWORKING
MOTU AVB networking has been designed to be powerful, yet straightforward to set up and use. Here are a few things that are useful to know.

Networking basics

- To make network connections, use CAT-5e or CAT-6 cables (a higher grade cable). They are available wherever network cables are sold.



- Network cable lengths can be long: 100 meters with standard copper wire cables; much longer with fiber-optic network cables.

Working with AVB switches

■ If you have MOTU AVB interfaces that are equipped with two network ports on them, you can daisy-chain up to eight devices in a single chain (with a total of seven connections in the chain).

■ For larger networks, use an AVB-compatible switch. You can use any standard AVB switch on the market. MOTU offers the six-port MOTU AVB Switch™ (sold separately).

☛ A non-AVB compatible switch will not work.

■ Connect MOTU AVB interfaces to any AVB Switch using their NETWORK ports.

■ You can also connect a Wi-Fi router, your Local Area Network (LAN), and/or your computer.

■ Expand the network by adding more switches. Make a single connection from one switch to the other.

■ You can daisy-chain devices and switches in serial fashion, but don't create loops. For example, in the network shown below, do not make any additional connections between any two switches.

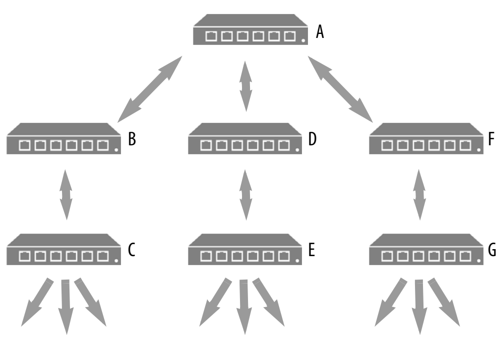


Figure 9-1: Use a hierarchical tree structure, as shown in this example. Be sure there is only one connection between any two devices on the network.

■ AVB audio may not pass reliably through more than seven switches, so plan your network topology accordingly to avoid this.

Working with computers on a network

■ Computers are not required for network operation, as you can control the network from iPads, tablets and smart phones.

■ To add computers to the network, connect them to any interface using Thunderbolt (which offers the highest possible channel counts). If Thunderbolt is not available, use USB.

■ A computer can be connected to the network through its Ethernet port, but only for the purposes of running the CueMix Pro app on the computer for command and control over the network. (In this scenario, you won't be able to stream audio to/from the network from the computer.)

■ All computers and interfaces on the network have full access to each other.

■ MOTU employs a 1 Gbit AVB implementation in the MOTU AVB Switch and 16A, allowing both devices to route many audio channels on the network.

SETTING UP THE 16A FOR NETWORKING

Once the 16A is connected to other AVB devices, either directly or via an AVB switch, using network cables as described earlier in this chapter, the 16A can stream audio to and from the other devices on the network.

The 16A supports sixteen AVB input streams and sixteen output streams. Each stream can be independently configured for one to eight audio channels.

16A input streams are used to receive streams broadcast by other connected AVB devices (*talkers*).

Conversely, 16A output streams broadcast to other AVB devices (*listeners*).

The 16A automatically "sees" the other connected AVB devices and allows you to make stream connections to and from them. Streams must be connected for audio to pass between the 16A and the other devices.

Connecting an input stream

To connect a 16A input stream to a stream from another device:

1 In the CueMix Pro app, go to the Device tab (page 33).

2 Choose any Input stream (it doesn't matter which one) and use the *Source Device* and *Source Stream* menus to specify the device and stream you wish to connect it to.

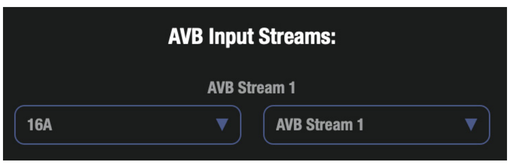


Figure 9-2: Connecting an input stream.

The input stream can now receive signal on audio channels from the other device.

☛ When connecting an input stream, the source device determines the number of audio channels in the stream and CueMix Pro will display the source channels accordingly in the Patchbay tab.

Connecting an output stream

To connect a 16A output stream to another device:

1 In the CueMix Pro app, go to the Device tab (page 33).

2 Choose any Output stream (it doesn't matter which one) and use the *Channels* and *Format* menus to specify the number of channels in the stream and its AVB stream format (see "AVB output stream format" below).

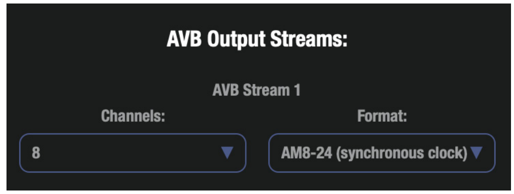


Figure 9-3: Connecting an output stream.

3 Click on the other device in the CueMix Pro sidebar to access its settings (item #10 on page 31).

4 Connect one of the device's input streams to the 16A output stream.

☛ Most AVB devices should display basic device and stream settings in CueMix Pro. In some cases, this may not be the case. If so, you will need to access the device's settings through the device itself, or its control software.

AVB output stream format

Since AVB was first developed, stream formats have evolved. *AM8-24* is an older format supported by earlier-generation MOTU AVB devices (such as the 1248, 8M, original 16A, etc.) and other 3rd-party AVB gear. *AAF-PCM* is a newer format that is supported by many Milan-compatible AVB devices. Below is a summary of which format to use in various situations.

Destination AVB device (listener)	16A output stream format to use
16A (2025)	AAF-PCM or AM8-24 (either works fine)
16A (2014)	AM8-24
Earlier-generation MOTU AVB interfaces	AM8-24
Older 3rd-party AVB devices	AM8-24
Both new and earlier MOTU AVB devices	AM8-24
Milan-compatible AVB devices	AAF-PCM

If you are not sure which format to choose, experiment with both until audio passes successfully between devices.

If you are sending a stream to both new and earlier-generation MOTU interfaces, use AM8-24. If you need to send the same audio signal to separate devices that don't support the same format, create two separate streams (for each format) and patch the audio signal to both streams (using the Patchbay).

Syncing AVB devices

When connecting AVB streams between two or more devices on a network, one device needs to be designated as the clock master, while all other devices use it as a clock source. Set the master device's Clock Source to *Internal*. Other devices should then resolve to the master device via an AVB stream connection.

For example, if you have two 16A's connected together via their network ports. set up their clocking as follows:

1 For 16A #1 (the unit connected to your computer), set its Clock Source to *Internal*.

2 For 16A #2 (the unit connected to #1 with a network cable), go to the Device tab *AVB Input Streams* section and choose 16A #1 as a source device for one of the streams and then choose a *Source Stream* from it (Figure 9-2 on page 62).

3 In the Device tab *Clock Source* menu, choose the AVB Input Stream you used above. Doing so resolves 16A #2 to 16A #1 via their AVB stream connection.

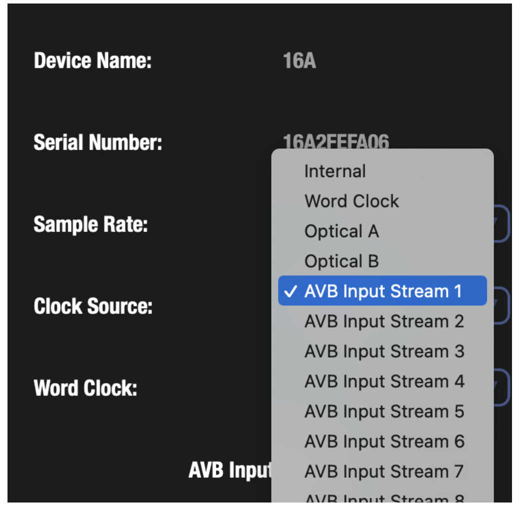


Figure 9-4: Choosing an AVB stream as a clock source.

For earlier-generation MOTU AVB interfaces, or 3rd-party AVB devices, you can access these same settings in the CueMix Pro Device tab, as described above. Just select the device in the sidebar (item #10 on page 31).

Making stream connections in the Patchbay

You can make stream connections in the Patchbay tab (page 36). Input streams are shown in the network SOURCES bank on the left; output streams are shown in the network DESTINATIONS bank on the right. Click the disclosure triangle for a stream to access the stream settings. These are the same settings as the Device tab, as described in the previous sections.

ROUTING AUDIO TO/FROM NETWORK STREAMS

Once you've made stream connections, as explained in the previous sections, use the Patchbay tab (page 36) to map input streams to destinations and to choose source audio channels for output streams.

Routing input streams

Input streams (coming from other devices) are shown in the network SOURCES bank on the left. Expand the stream to view its channels and use the patchbay to connect those stream channels to any desired destination channel in the right-hand bank. Possible destinations include the computer, the 16A mixer, physical 16A outputs (analog or optical), and even output AVB streams.

Choosing sources audio channels for output streams

16A output streams being broadcast to other AVB devices are listed in the DESTINATIONS bank on the right of the Patchbay tab. Expand the stream to view its channels and use the patchbay to connect source audio channels to the stream's audio channels. Possible sources include 16A physical inputs (analog and optical), computer output channels, 16A mixer bus outputs, and even input AVB streams. You can also specify the number of channels in the stream, and the stream format (explained earlier).

MAPPING COMPUTER CHANNELS TO NETWORK STREAMS

If a host computer is connected to an interface (through Thunderbolt or USB), mapping network input and output streams is accomplished as described in the previous two sections. Simply enable AVB streams as desired, and map them to computer channels in the Patchbay tab.

DEVICE PRESETS AND AVB STREAM CONNECTIONS

When you save a 16A preset, any AVB stream connections that it has established with other devices on the network are included with the saved preset. When you recall the preset, those saved stream connections are restored, as long as the other devices are still present on the network and broadcasting (and listening to) the same streams as when the preset was saved. If the other device is not present (or perhaps turned off), its streams will be reported as *offline*.

In general, if you have multiple devices on a network with interconnecting AVB streams, and you wish to preserve the state of the network, it is recommended that you save a device preset for each device on the network. Doing so will allow you to faithfully restore the entire network stream configuration by recalling each device's saved preset.

Part 3

Appendices

APPENDIX A Troubleshooting

My 16A isn't showing up in Audio MIDI Setup on my Mac.
Due to the updated architecture of macOS Catalina (10.15) and above, the system extensions for all newly-installed third-party software will automatically be blocked from running. If your 16A is not showing up in Audio MIDI Setup, CueMix Pro or your DAW, you might need to enable the driver in your System Preferences. To do so, first download and install the very latest installer for your 16A. After restarting, open System Preferences. Select Security & Privacy. In the General section, click the Allow button. The Allow button will disappear 30 minutes after installation. If the button has disappeared, run the installer for the MOTU driver again.

I have absolutely no audio input or output happening to or from my 16A. Why?
Make sure that the unit has a stable sample rate (the *lock icon* in the front panel display will turn red if the clock hasn't settled yet). Try setting the unit's clock source to *Internal* if you can't sync to any external clock sources. Check that audio is working with Internal sync, and if so, then work on establishing a stable external clock.

I can't hear computer audio output through my 16A.
In the Sound panel of System Preferences, the 16A should be selected as the output device.

How do I monitor live 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 mixer in the 16A. Please see "Monitoring through the 16A" on page 51.

How do I control monitoring latency?
See "Reducing monitoring latency" on page 51.

How do I factory reset my device?
Push SELECT or turn the MENU knob to enter the main menu. Navigate to *RESET* and tap the SELECT button three times to reset.

I hear clicks and pops under optical or S/PDIF sync.
Many problems result from incorrect sync settings. See "Synchronization" on page 22. Whenever there is any unexpected noise or distortion, suspect digital clocking issues.

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

CUSTOMER SUPPORT

We are happy to provide complimentary customer support to our registered users. If you haven't already done so, please take a moment to register online at MOTU.com. Doing so entitles you to technical support and notices about new products and software updates.

TECHNICAL SUPPORT

If you are unable, with your dealer's help, to solve problems you encounter with your MOTU device, you may contact our technical support department in one of the following ways at motu.com/support:

- Live Chat: You can connect directly with a technician Monday through Friday between 10 AM and 5 PM Eastern Time.
- Schedule a call: You can schedule a callback time to speak with a technician by phone Monday through Friday between 1 PM - 5 PM Eastern time.

- Support ticket: You can submit an online support ticket at any time, 24 hours a day, 7 days a week. A technician will get back to you in 1-2 business days.

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

- The serial number of your MOTU device. This is printed on a label placed on the bottom of the unit and on the side of the box. 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 that refer to the features or operation of your MOTU Device or Performer Lite 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 support ticket or chat 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 Development Team, MOTU Inc., 1280 Massachusetts Avenue, Cambridge, MA 02138, or use our online suggestion box at www.motu.com/suggestions.

APPENDIX B Audio Specifications

Line In

Connector Type	1/4" Female, TRS	Balanced/unbalanced, Tip hot
Specification	Complies with EBU-R68 / SMPTE RP-155	
Impedance Load	20 k ohm	balanced
Dynamic Range	120 dB	A-weighted
THD+N	-114 dB	1kHz, 20dBu, 0dB Trim, unweighted
Frequency Response	±0.05 dB, 20 Hz-20 kHz	Ref. 1 kHz
Max Level In	+21 dBu	
Gain Range	0 to +20 dB	Digitally controlled in 1 dB steps

Line Out

Connector Type	1/4" Female, TRS	Balanced, tip hot
Output Impedance	220 ohm	Per leg
Dynamic Range	125 dB	A-weighted
THD+N	-114 dB	-1 dBFS, Unweighted, 1 kHz
Frequency Response	+0.01/-0.15 dB, 20 Hz-20 kHz	Ref. 1 kHz
Max Level Out	+21 dBu	
Trim Range	0 to -99 or -∞	Digitally controlled in 1 dB steps

Phones

Connector Type	1/4" Female, TRS Stereo	Tip Left, Ring Right
Output Impedance	< 1 ohm	
Dynamic Range	118 dB	
THD+N	-110 dB	Unweighted, -1 dBFs, 1 kHz
Frequency Response	+0.01/-0.1 dB, 22 Hz-20 kHz	Ref. 1 kHz
Trim Range	0 to -99 dB, plus mute	0 to -99 dB in 1 dB steps
Max output	13.3 dBu	Unloaded

Word Clock In/Out/Thru

Specification	AES-11 2009 Annex B	
Connector Type	BNC	
Termination	75 ohm (in/out)	THRU is unterminated
Lock Range	44.1 kHz / 48kHz, +- 0.5%	x1/x2/x4
Input	1 vpp to 3 v p-p (with termination)	AC coupled
Output	5.0 vpp, (2.5 v p-p terminated)	DC coupled
Jitter	complies with AES3-4-2009	< 0.025 UI

Power Supply

Connector Type	IEC 3-conductor receptacle	For AC mains connection
Configuration	Internal, Universal	
Power Input	100 V to 240 V, 50 Hz or 60 Hz	
Power Usage	0.5A	