

# **Track16™**

## **User Guide for Windows**

**MOTU**

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

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

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

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

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



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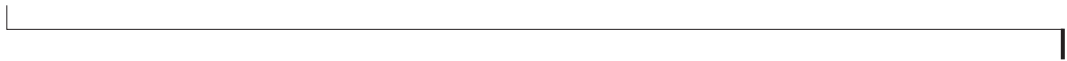
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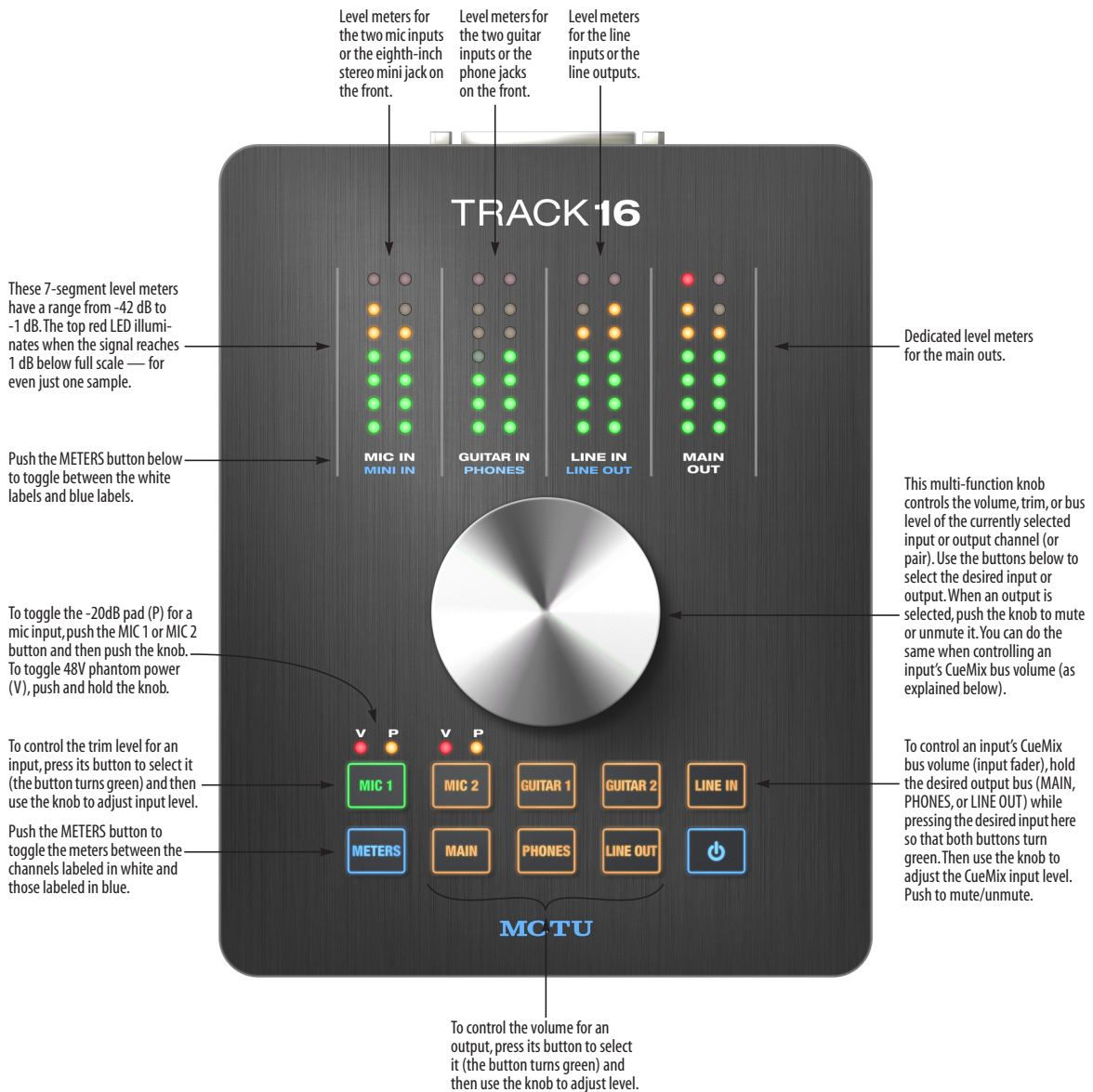


*Part 1*

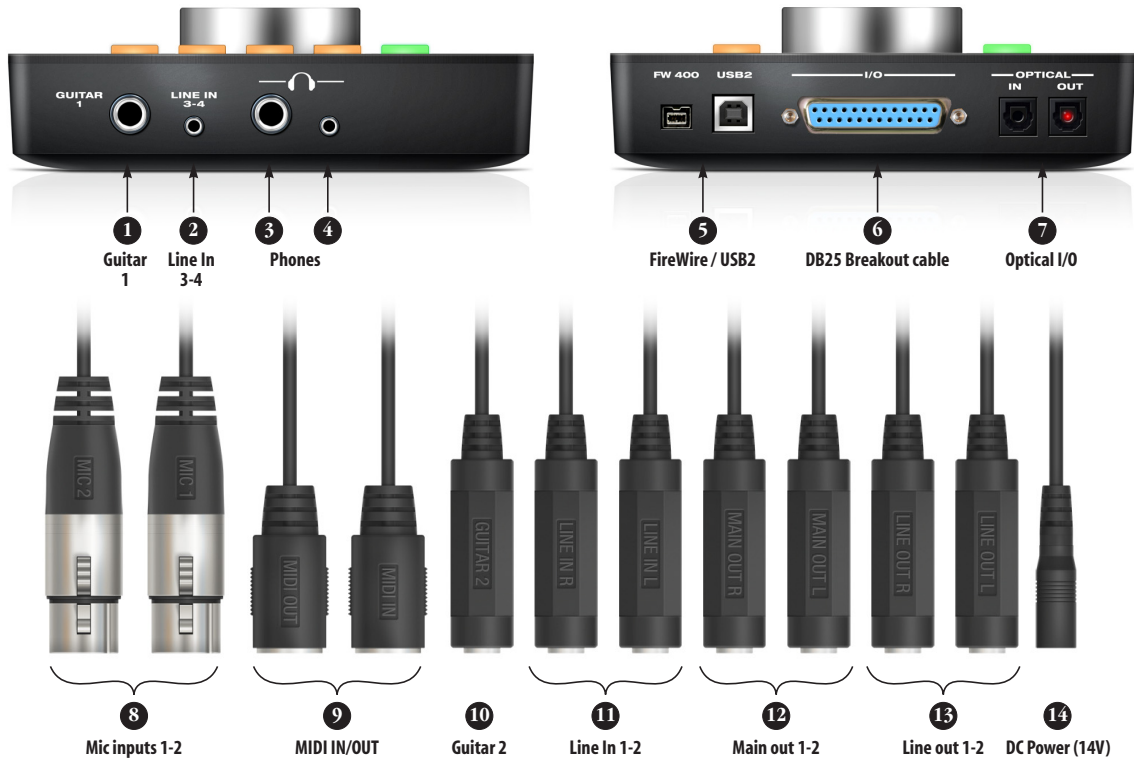
*Getting Started*



# Quick Reference: Track16 Operation



# Quick Reference: Track16 Connectors



1. This is one of two hi-Z guitar inputs. The second guitar input is on the breakout cable (10).
2. Line In 3-4 is a stereo eighth-inch "mini" jack for connecting an iPod or other mobile audio device.
3. From the factory, the headphone jack is a discrete output, but it can mirror any other output pair (digital or analog).
4. If your headphones have a stereo mini connector, connect them here. The signal on the two headphone jacks (TRS and stereo mini) is the same.
5. Connect the Track16 to the computer here via either FireWire or USB2, using either the standard 1394 FireWire B or USB cable provided with your Track16. There's not much difference, except that FireWire offers bus-powered operation (without the DC power supply). **Important note: it is best to turn off the Track16 when plugging in the FireWire cable, as this avoids the possibility of static discharge, which can harm the electrical components in the Track16 or your computer.**
6. Connect the breakout cable supplied with your Track16 here. For optimum performance, use only the supplied Track16 cable.
7. These optical digital I/O connectors can be connected either to an ADAT-compatible "lightpipe" device (such as a digital mixer) or to a S/PDIF optical ("TOSLINK") compatible device, such as an effects processor. Be sure to set the format in the MOTU Audio Console software. (See "Optical input/output" on page 36 for details.) ADAT optical supplies eight channels of 24-bit digital I/O per bank (4 channels per bank at 96kHz). TOSLINK is stereo at sample rates up to 96 kHz.
8. These XLR connectors accept a mic cable and are equipped with a preamp, 48V phantom power and 20 dB pad can be applied via the controls on the top panel. The knob provides 60 dB of gain.
9. Connect a MIDI device here using standard MIDI cables. Connect the Track16's MIDI OUT port to the MIDI IN port on the other device. Conversely, connect the Track16's MIDI IN port to the MIDI OUT port on the other device. You can connect different devices to each port, such as a controller device to the IN port and a sound module to the OUT port. You can also daisy-chain MIDI devices, but be sure to manage their MIDI channels (so that they don't receive or transmit on the same channel).
10. This is the second of two hi-Z guitar inputs. The first guitar input is on the front panel (1).
11. These two analog line-level inputs are balanced quarter-inch connectors. They do not have preamps, so they are best used for synthesizers, drum machines, effects processors, and other instruments with line level signals (either -10 dB or +4 dB). These inputs are also equipped with the Track16's Precision Digital Trim™ feature: digitally controlled analog trims that let you adjust input level in 1/4 dB increments. The trim can be adjusted over a range of -96 to +22 dB.
12. These two balanced TRS jacks serve as the Track16's main analog outputs. You can connect them to a set of powered studio monitors and then control the volume by pressing the MAIN button.  
  
To hear disk tracks in your audio software on these main outs, assign the disk tracks (and master fader) to these main outs. Also make sure the Main Out Assign option is set to *Main Out 1-2*. See "Main Out Assign" on page 36. You can also use CueMix FX to monitor live Track16 inputs here as well.
13. These two analog line outputs are unbalanced quarter-inch connectors referenced to +17 dBu.
14. This jack accepts any standard 9-18V DC power supply with either tip-positive or tip-negative polarity. Alternately, you can use FireWire bus power.



# Quick Reference: MOTU Audio Console

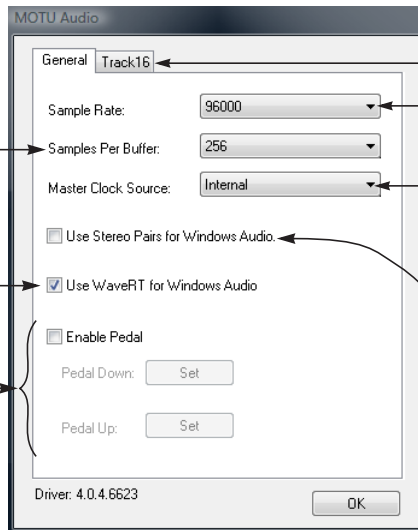
Choosing a smaller setting here reduces the delay you may hear when listening to live input that you are running through effects plug-ins in your software. But lower settings also increase the strain on your computer. For details, see “Samples Per Buffer” on page 35.

Uncheck this option if the Windows audio software you are using with the Track 16 does not support Windows WaveRT drivers and instead only supports WDM drivers.

These foot switch settings are for other MOTU audio interfaces and don't apply to the Track 16.

This menu lets you choose what you will hear from the PHONES jack. To mirror the main outs, choose *Main Out 1-2*. Or you can mirror any other output pair. To hear the phones as their own independent output, choose *Phones 1-2*.

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

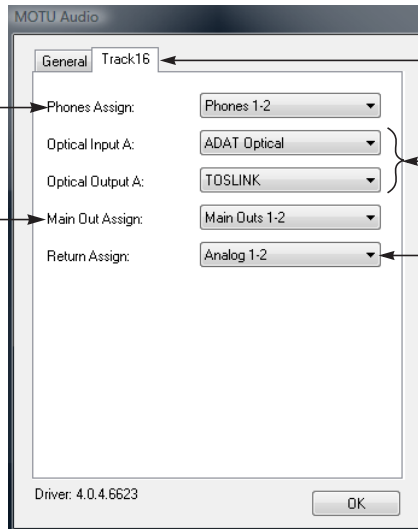


Click the tabs to access general MOTU interface settings or settings specific to the Track 16 (or other connected interface.)

Choose the global sample rate for the system here.

Determines the clock source for your Track 16. If you're just using the analog ins and outs, set this to 'Internal'. The other settings are for digital transfers via the optical ports or for resolving the Track 16 to other clock sources.

This option should always be left on (checked). There are only a few rare cases in which you would want to turn it off.



Click the Track 16 tab to access these settings.

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

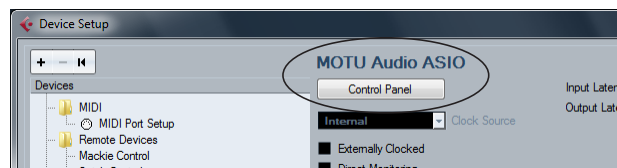
The Track 16 driver provides a stereo return back to the computer. This return feeds the signal on any Track 16 output pair directly back to the computer, where you can record, process, monitor or otherwise use it. This is a great way to “bounce” full mixes, complete with live audio routed through the Track 16 only, back into the computer.

## How to access these settings

There are several ways to access these settings:

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

MOTU Audio Console in Cubase





# CHAPTER 1 About the Track16

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## OVERVIEW

The Track16 is a hybrid FireWire USB2 audio interface for Mac and Windows that provides 16 inputs and 16 outputs (at 44.1 or 48kHz). Both analog and digital I/O are offered at sample rates up to 96 kHz, and analog recording and playback is offered at rates up to 192 kHz. All inputs and outputs can be accessed simultaneously. The Track16 consists of a rugged, aluminum alloy desktop unit that connects to a computer via a standard FireWire or USB cable.

The Track16 offers the following main features:

- Universal computer connectivity via FireWire or high-speed USB2
- Two analog mic inputs equipped with independent mic preamps
- Two hi-Z analog guitar inputs
- Two balanced analog line inputs on quarter-inch connectors
- Stereo unbalanced analog line input on an 1/8th-inch “mini” connector
- Two analog main outputs on balanced quarter-inch TRS connectors
- Two analog line level outputs on unbalanced quarter-inch connectors

- Independent 48V phantom power and 20 dB pad for each mic input
- Independent Precision Digital Trim™ for each input
- 24-bit operation on all analog I/O at standard sample rates up to 192 kHz
- One bank of optical digital I/O that provide 8 channels of ADAT optical at 48 kHz, 4 channels of S/MUX optical I/O at 96 kHz, or stereo TOSLINK at rates up to 96 kHz
- On-board SMPTE synchronization via any analog input and output
- Two phone jacks (quarter-inch and mini)
- Programmable master volume knob
- CueMix™ FX no-latency mixing, monitoring and effects processing
- 7-segment LED level meters for all analog inputs
- Dedicated 7-segment level meters for main outs
- External international power supply
- FireWire bus powered operation
- Stand-alone operation
- Mac and Windows drivers for multi-channel operation and across-the-board compatibility with any audio software on current Mac and Windows systems

With a variety of I/O formats, mic preamps, no-latency mixing and processing of live input and synchronization capabilities, the Track16 is a complete, portable “studio in a box” when used with a Mac or Windows computer.

## THE TRACK16 I/O CONNECTIONS

The Track16 has the following I/O connections, supplied either on the base unit itself or on the included DB25 breakout cable. All Track16 inputs and outputs can be used simultaneously, for a total of 16 inputs and 16 outputs at 44.1/48kHz:

Connection	Input	Output
Mic 24-bit 192 kHz on XLR with preamp	2	-
Guitar 24-bit 192 kHz on hi-Z quarter-inch	2	-
Analog 24-bit 192 kHz on bal/unbal TRS	2	4
Analog 24-bit 192 kHz on unbal stereo "mini"	2	0
Headphone output*	-	4
ADAT optical digital†	8	8
<b>Total</b>	<b>16</b>	<b>16</b>

\* The two phone outputs (quarter-inch and mini) are hard-wired to mirror each other. Their signal can be an independent output stream, or it can mirror any other Track16 output pair, such as the main outs.

† The Track16 optical connectors support several standard optical I/O formats, which provide varying channel counts. See "Optical input/output" on page 36 for details about optical bank operation.

With the exception of the phone jacks on the front panel, all inputs and outputs are discrete (independent).

### Analog

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

### Mic inputs

Both mic inputs are equipped with a separate preamp. Individual 48V phantom power and 20 dB pad are supplied by the top panel switches. Each input can be trimmed with the volume knob.

### Guitar inputs

Without proper impedance levels, direct "DI" guitar input won't sound right when you run it through your guitar amp plug-ins and virtual stomp boxes. The Track16's dedicated "hi-Z" guitar inputs provide high-impedance load characteristics suitable for passive and active pickups alike, to produce authentic guitar tone and feel with your guitar processing software.

### Line inputs and outputs

All quarter-inch analog connectors are balanced, except for line outputs 3-4. Balanced connectors can accept either a balanced or unbalanced plug. Balanced output connections protect audio signals from RF interference, AC hum, and other noise-producing environmental factors. The unbalanced line outputs are useful for home stereo components, consumer speakers, or other common unbalanced destinations.

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

### Precision Digital Trim™

All of the Track16's analog inputs are equipped with digitally controlled analog trims, adjustable in approximately 1/4 dB increments. All analog inputs can be trimmed with the Track16 knob or with the Track16's included CueMix FX control software for Mac and Windows. This gives you finely-tuned control of trim settings for a wide variety of analog inputs for optimum levels. Different trim configurations can then be saved as preset configurations for instant recall.

### Optical

The optical connectors provide 8 channels of ADAT optical at 44.1 or 48 kHz, 4 channels of S/MUX optical I/O at 96 kHz or stereo TOSLINK at rates up to 96 kHz. The banks operate indepen-

dently, allowing you to mix and match any optical formats on input and output. For example, you could receive 4 channels of 96 kHz S/MUX input while at the same time sending 96 kHz stereo optical S/PDIF (“TOSLINK”) output.

### **MIDI I/O**

The Track16’s standard MIDI IN and MIDI OUT jacks supply 16 channels of MIDI I/O to and from the computer.

### **On-board SMPTE synchronization**

The Track16 can resolve directly to SMPTE time code via any analog line input, without a separate synchronizer. The Track16 can also generate SMPTE time code via any analog output. The Track16 provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy.

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

### **Hybrid FireWire/USB2 connectivity**

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

To fully support both formats, the Track16 audio interface is equipped with both FireWire B (400 Mbit/sec) connectors and a high-speed USB2 (480 Mbit/sec) connector, and you can use either port to connect the Track16 to your computer. This gives

you maximum flexibility and compatibility with today’s ever-expanding universe of Mac and Windows computers.

### **Bus-powered operation**

When connecting to the computer with FireWire, the Track16 can draw power from the FireWire connection itself, without the Track16 power adapter. For more information about bus powered operation, see “Power options” on page 26.

## **THE TRACK16 TOP PANEL**

### **Metering**

The top panel of the Track16 displays four pairs of 7-segment ladder LEDs. Push the METERS button to toggle between the channels indicated by the white labels and channels indicated by the blue labels. The right-hand meters are dedicated to the main outs.

### **Multifunction volume knob with buttons**

The multifunction volume knob controls the volume for the currently selected input or output, as chosen by the buttons below. Simply push the button for the input or output you wish to control.

The multifunction knob can also be pushed to mute and unmute the current channel. For the two mic inputs, it can toggle pad and 48V phantom power.

It is also possible to control input mix levels with the volume knob.

For a summary of all knob functions, see “Quick Reference: Track16 Operation” on page 7.

### **16-BIT AND 24-BIT RECORDING**

The Track16 system handles all data with a 24-bit signal path, regardless of the I/O format. You can record and play back 16-bit or 24-bit audio files at any supported sample rate via any of the Track16’s

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

### **CUEMIX FX 32-BIT FLOATING POINT MIXING AND EFFECTS**

All Track16 inputs and outputs can be routed to the on-board CueMix FX 8-bus (4 stereo) digital mixer driven by hardware-based DSP with 32-bit floating point precision. The mixer allows you to apply no-latency effects processing to inputs, outputs or busses directly in the Track16 hardware, independent of the computer. Effects can even be applied when the Track16 is operating stand-alone (without a computer) as a complete rack-mounted mixer. Input signals to the computer can be recorded wet, dry, or dry with a wet monitor mix (for musicians during recording, for example).

Effects include reverb, parametric EQ and compression/limiting. The Track16's Classic Reverb™ provides five different room types, three frequency bands with adjustable crossover points, shelf filtering and reverb lengths up to 60-seconds.

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

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

The Track16's flexible effects architecture allows you to apply EQ and compression on every input and output (a total of 30 channels), with enough

DSP resources for at least one band of parametric EQ and compression on every channel at 48 kHz. However, DSP resources are allocated dynamically and a DSP meter in the CueMix FX software allows you to keep tabs on the Track16's processing resources. Each input, output and mix bus provides a send to the Classic Reverb processor, which then feeds reverb returns to mix busses and outputs, with a selectable split point between them to prevent send/return feedback loops.

### **INCLUDED SOFTWARE**

The Track16 software installer provides the following including cross-platform software applications for Mac and Windows.

#### **MOTU Audio Console**

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

#### **MOTU SMPTE Console**

MOTU SMPTE Console provides access to the Track16 system's SMPTE time code synchronization features, including locking to time code, generating time code, displaying a time code read-out, and so on.

#### **CueMix FX**

CueMix FX gives you complete control over the Track16's CueMix FX on-board mixer, which provides no-latency monitoring, mixing and processing of live inputs through your Track16.

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

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

### **HOST AUDIO SOFTWARE**

The Track16 system ships with Windows drivers that allows you to record, edit, play back and mix your Track16 projects using your favorite Windows audio software.





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

### **PACKING LIST**

The Track16 ships with the items listed below. If any of these items are not present in your Track16 box when you first open it, please immediately contact your dealer or MOTU.

- One Track16
- One Track16 DB25 I/O cable
- One 9-pin to 9-pin IEEE 1394 “FireWire” B cable
- One USB cable
- One universal DC power adapter
- One Track16 Mac/Windows manual
- One cross-platform installer disc
- Product registration card

### **WINDOWS SYSTEM REQUIREMENTS**

The Track16 system requires the following Windows system:

- A 1 GHz Pentium-based PC compatible or faster equipped with at least one USB2 or FireWire port
- 1 GB RAM; 2 GB or more recommended
- Windows 7 or Vista, 32- or 64-bit; Vista SP 2 or later required
- Available FireWire or high-speed USB 2.0 port
- A large hard drive (preferably at least 250 GB)

### **PLEASE REGISTER TODAY!**

Please register your Track16 today. There are two ways to register.

- Visit [www.motu.com/register](http://www.motu.com/register)

OR

- Fill out and mail the included product registration card

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

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



# CHAPTER 3 Installing the Track16 Software

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## INSTALLATION

Before you connect the Track16 to your computer and power it on, run the Track16 software installer. This ensures that all the Track16 components are properly installed in your system.

☛ If you've already connected the Track16 to your computer and powered it on, Windows may issue an alert notifying you that the Track16 requires drivers, followed by another window asking you to locate the drivers. If this happens:

- 1 Cancel the driver search.

- 2 Power off and disconnect the Track16.

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

### Run the MOTU Audio installer

Install the Track16 software as follows:

- 1 Insert the MOTU Audio Installer disc; or, if you have downloaded the MOTU Audio installer, locate the folder containing the download.

- 2 Read the *Read Me* file for installation assistance and other important information.

- 3 Open the *Setup Audio* application.

- 4 Follow the directions that the installer gives you.

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

Software component	Purpose	For more information
MOTU Audio and MIDI drivers	Provides multi-channel audio input and output for MOTU FireWire and USB Audio devices with host audio software.	"MOTU Audio drivers" on page 20
MOTU Audio Console	Provides access to all of the settings in the Track16 and other MOTU interfaces. Required for Track16 operation.	chapter 5, "MOTU Audio Console" (page 33)
CueMix FX	Gives you complete control over the Track16's CueMix FX on-board mixer, which provides no-latency monitoring, mixing and processing of live inputs through your Track16.	chapter 9, "CueMix FX" (page 55)
MOTU SMPTE Console	Provides access to the Track16 system's SMPTE time code sync features.	chapter 10, "MOTU SMPTE Console" (page 101)

## MOTU AUDIO DRIVERS

### ASIO

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

For details about using the Track16 with ASIO, see chapter 7, “Configuring Host Audio Software” (page 41).

### WDM

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

For details about using the Track16 with WDM, see chapter 7, “Configuring Host Audio Software” (page 41).

### WaveRT

Uncheck this option (Figure 5-1 on page 34) if the Windows audio software you are using with your MOTU audio interface does not support Windows WaveRT drivers and instead only supports WDM drivers.

## MOTU AUDIO CONSOLE

MOTU Audio Console (available in the Start menu) gives you access to all of the settings in the Track16, such as the clock source and sample rate. For complete details, see chapter 5, “MOTU Audio Console” (page 33).

## MOTU MIDI DRIVER

This driver allows you to access the Track16’s MIDI input and output ports. The ports are published in Windows and are available to all MIDI software.

## CUEMIX FX

CueMix FX (available in the Start menu) provides control over the Track16’s no-latency CueMix FX on-board mixing, effects processing, an instrument tuner, a full-featured oscilloscope, and other audio analysis tools. For details, see chapter 9, “CueMix FX” (page 55).

## MOTU SMPTE CONSOLE

MOTU SMPTE Console (available in the Start menu) software provides a complete set of tools to resolve the Track16 to SMPTE time code, and to generate SMPTE for striping, regenerating or slaving other devices to the computer. For details, see chapter 10, “MOTU SMPTE Console” (page 101).

# CHAPTER 4 Installing the Track16 Hardware

## OVERVIEW

Here's an overview for installing the Track16:

<b>Connect the Track16 interface</b> .....	21
Connect the Track16 to the computer.	
<b>Connect audio inputs and outputs</b> .....	23
Make optical and analog connections as desired.	
<b>Connect MIDI gear</b> .....	26
Connect a controller, synth or control surface.	
<b>Power options</b> .....	26
Choose from among several convenient options.	
<b>A typical Track16 setup</b> .....	28
An example setup for computer-based mixing/FX.	
<b>Connecting multiple MOTU FireWire interfaces</b> .....	29

## CONNECT THE TRACK16 INTERFACE

Your Track16 audio interface is equipped with both FireWire B connectors (running at 400 Mbit/sec) and a high-speed USB2 connector (480 Mbit/sec), and you can use either port to connect the Track16 to your computer. This gives you maximum flexibility and compatibility with today's ever-expanding universe of Mac and Windows computers.

### Type B FireWire ports

The Track16 has a FireWire 9-pin Type B port, which provides the most reliable FireWire connection available. The port operates at 400 Mbit/s, and it can be connected to any available FireWire port on your computer: Type A (6-pin), Type A "mini" (4-pin), or Type B (9-pin). If your computer has FireWire Type B ports, use the included 9-pin-to-9-pin FireWire cable. If your computer has either standard Type A ports or

miniature Type A ports, use the appropriate 9-pin-to-6-pin or 9-pin-to-4-pin FireWire cable (sold separately).

### Which should I use: FireWire or USB2?

If your computer does not have a FireWire port, then obviously you will need to connect the Track16 to one of its high-speed USB 2.0 ports.

If your computer has both FireWire and USB2, then it is your choice, and your decision may depend mostly on other peripherals you may also have, or bus power, as explained below.

### Bus-powered operation requires FireWire

There is only one significant difference between FireWire and USB2 operation: bus power. FireWire provides enough power on the FireWire bus that the Track16 can be powered solely by its FireWire connection to the computer. For complete information, see "Bus power requirements" on page 26.

If you use USB2, you must also use the DC power supply included with your Track16, as USB2 does not supply enough power by itself.


### If you are connecting via FireWire

**1** Before you begin, make sure your computer and the Track16 are switched off.

**2** Plug one end of the Track16 FireWire cable (included) into the FireWire socket on the computer.

☛ You can also connect the Track16 to a 400Mbit "FireWire A" port using a 9-pin-to-6-pin FireWire B cable (not included). The Track16 will still operate at its specified 400Mbit (FireWire A) data rate.

**3** Plug the other end of the FireWire cable into the Track16.

 Make absolute sure to align the notched side of the FireWire plug properly with the notched side of the FireWire socket on the Track16. If you attempt to force the plug into the socket the wrong way, you can damage the Track16.

### High Speed USB 2.0 versus USB 1.1

There are primarily two types of USB host controllers widely available on current personal computers. USB 1.1 controllers support simple peripherals that don't require a high speed connection, such as a computer keyboard, a mouse, or a printer. USB 2.0 controllers support high speed devices such as the Track16. Since the Track16 requires a high speed connection, it must be connected to a USB 2.0 host controller or hub.

For the most reliable connection, it is recommended that you connect the Track16 directly to one of your computer's USB 2.0-compatible ports. However, since USB 2.0 hubs are compatible with both types of devices, the Track16 can be connected to a USB 2.0 hub along with USB 1.1 devices if necessary. The Track16 will not operate properly if it is connected to a USB 1.1 hub.

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

- 1** Open the Windows Control Panel, and choose Hardware and Sound.
- 2** In the Devices and Printers section, click Device Manager.
- 3** In Device Manager, open the Universal Serial Bus Controllers section.
- 4** Look in the list of USB devices. An Enhanced USB Host Controller Interface (EHCI) represents a USB 2.0 controller. An Open USB Host Controller

Interface (OHCI) or Universal USB Host Controller Interface (UHCI) represents a USB 1.1 controller.

### If you are connecting via high-speed USB 2.0

- 1** Before you begin, make sure your computer and the Track16 are switched off.
- 2** Plug the flat "type A" plug of the Track16 USB cable (included) into a USB2-equipped socket on the computer.
- 3** Plug the squared "type B" plug of the USB cable into the Track16 I/O.

### Switching Between FireWire and USB

Most of the time, the Track16 can detect whether a FireWire or a USB cable is connected to it, and it will automatically switch to FireWire operation or USB operation accordingly. Some FireWire or USB ports that do not provide bus power cannot be automatically detected by the Track16, and you may need to switch the device to FireWire mode or USB mode manually:

To switch to	Do this
FireWire	Hold down the MIC 1 button while powering up
USB	Hold down the MIC 2 button while powering up

When the Track16 switches to FireWire operation, it displays *F* on its meter LEDs. When it switches to USB operation, it displays *U* on the meter LEDs.

## CONNECT AUDIO INPUTS AND OUTPUTS

Track16 connectors are located on the front and rear of the base unit. The DB25 connector on the rear panel connects to the Track16's included breakout cable, which provides further connections for audio, MIDI and power.

The base unit (Figure 4-1) provides the following connectors:

- Guitar In 1
- Line In 3-4 (1/8th-inch stereo mini)
- Phones
- Optical In and Out
- FireWire
- USB2
- DB25

The breakout cable (Figure 4-3) provides the following connectors:

- Mic In 1-2 (XLR with preamp, 48V and pad)
- Guitar In 2 (hi-Z)
- Main Out 1-2 (quarter-inch, balanced)
- Line Out 3-4 (quarter-inch, unbalanced)
- Line In 1-2 (quarter-inch, balanced)
- MIDI In and Out
- DC power (for the included DC power adapter)

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

## Track16 breakout cable

The breakout cable included in the Track16 package (Figure 4-3) is a custom-engineered cable with special shielding, twisted pairs, and other characteristics designed to ensure the best possible audio quality and performance. Attach it to the DB25 connector on the Track16 base unit and fully tighten the screws to ensure a secure connection.

You can operate the Track16 without the breakout cable if both of these conditions are true:

- The Track16 is connected to a computer with a FireWire cable for bus-powered operation, and
- you plan to use only the I/O connectors on the base unit itself (guitar, phones, and/or optical).

## Mic inputs 1-2

Using a standard mic cable, connect microphones to the female XLR connectors on the breakout cable labeled *Mic In 1* and *Mic In 2*.

## Phantom power

If you are connecting a condenser mic or other device that requires phantom power, you can enable (or disable) phantom power as follows:

- 1 Make sure the microphone is plugged in first.
- 2 Power on the Track16.
- 3 Push the corresponding *MIC 1* or *MIC 2* button on the Track16 to select the input (Figure 4-2).
- 4 Push and hold the main knob for a few seconds.



Figure 4-1: the Track16 front and rear panels.

- 5 The red “V” LED illuminates (Figure 4-2).
- 6 To disable phantom power, simply repeat this procedure. The red “V” LED turns off.

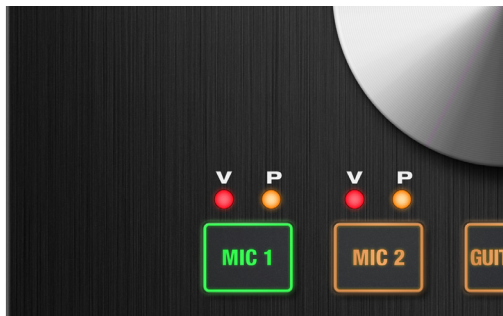


Figure 4-2: The phantom power (V) and pad (P) LEDs for the mic inputs

### Trim

Each mic input is equipped with preamp that delivers 60 dB gain. Push the MIC 1 or 2 button and then turn the knob to adjust the input level as needed for each input. Use the mic level meters to monitor the current input level.

The Track16’s input trims are digitally controlled, so they allow you to make fine-tuned adjustments in approximately 1 dB increments. You can also adjust trim in the MOTU CueMix FX software. See “Input trim” on page 61.

### -20 dB pad

If the input signal is too hot with the trim turned all the way down, engage the -20dB pad for the input. To toggle the pad, push the appropriate mic button and then quickly push the knob. The amber “P” LED will turn on or off accordingly.

### Muting mic inputs

Mic inputs without a mic or terminator connected can pick up noise, so keep them muted when not in use. See “Muting/unmuting” on page 40.

### Guitar inputs

The Track16’s dedicated “hi-Z” guitar inputs provide high-impedance load characteristics suitable for passive and active pickups alike, to produce authentic guitar tone and feel with your guitar processing software.

### Line inputs

The Track16 provides two stereo line inputs:

- Line In 1-2 (quarter-inch, balanced) on the breakout cable
- Line In 3-4 (1/8th-inch stereo mini) on the front panel of the base unit

These inputs can be used with any line level source, such as a keyboard, drum machine, outboard processor, iPod, or iPad.

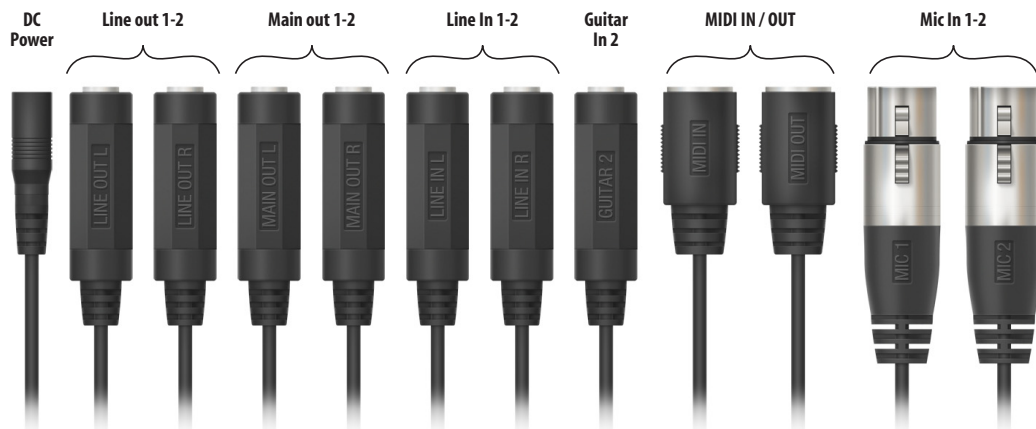


Figure 4-3: the Track16 breakout cable.



## Line input trims

The line inputs are calibrated to accommodate either +4 dB or -10 dB signals and are equipped with digitally controlled analog trims that provide -96 dB of cut and +22 dB of gain. You can use either the main volume knob or the included CueMix FX software to adjust the input trim. To adjust these trims using CueMix FX, see “Input trim” on page 61. To adjust the trims using the volume knob:

- 1 Push the corresponding input button.
- 2 Turn the volume knob.

Use the level meters to observe the current gain.

## Main outs and line outs

The stereo quarter-inch main outputs are balanced line-level outputs that also accept an unbalanced plug. The line outputs are unbalanced. All four outputs are calibrated to produce a +4 dBu line level output signal, according to the EBU-R68 specification, as follows:

EBU-R68 measurement	dBVU	dBu	dBFS
Meter alignment	0	0	-18
Peak level	+8	+8	-10
Headroom	10 dB typical peak to clip		

In a standard studio configuration, the main outs are intended for a pair of studio monitors, but they can also be used as additional outputs for any purpose.

## Optical

The Track16 rear panel provides one set of ADAT optical (“lightpipe”) connectors (Figure 4-1). These optical in/out connectors can operate independently and offer two different optical formats: ADAT optical or TOSLink (optical S/PDIF). For example, you could connect

8-channel ADAT optical input from your digital mixer and stereo TOSLink output to an effects processor.

## Optical trim

The Track16 supplies +12dB of digital trim (boost) for the optical input, which can be adjusted from CueMix FX (“Input trim” on page 61).

## Optical operation at 44.1 or 48 kHz

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

## ADAT optical operation at 88.2 or 96 kHz

When configured for ADAT “lightpipe”, an optical connector provides four channels at 88.2 or 96 kHz (2x sample rates) using the industry standard SMUX high-resolution optical format. If you are connecting a MOTU interface with optical ports to the Track16, use the *Type I* SMUX setting in the other MOTU interface. Refer to the documentation for your other MOTU interface for details on how to make this setting.

Below is a summary of optical formats:

Format	44.1 or 48 kHz	88.2 or 96 kHz
ADAT optical	8 channels	4 channels of SMUX
TOSLink	stereo	stereo

## Choosing a clock source for optical connections

When connecting an *optical* device, make sure that its digital audio clock is phase-locked (in sync with) the Track16. There are two ways to do this:

1. Resolve the optical device to the Track16
2. Resolve the Track16 to the optical device

For 1), choose *Internal* (or any other clock source except *ADAT optical*) as the clock source for the Track16 in MOTU Audio Setup.

For 2), choose *ADAT Optical* as the Track16’s clock source.

For details about using the clock source setting and the MOTU Audio Setup software in general, see chapter 5, “MOTU Audio Console” (page 33).

## CONNECT MIDI GEAR

Connect your MIDI device’s MIDI IN jack to the Track16’s breakout cable MIDI OUT jack (Connection A below). Conversely, connect the MIDI device’s MIDI OUT jack to the Track16’s MIDI IN jack (Connection B).

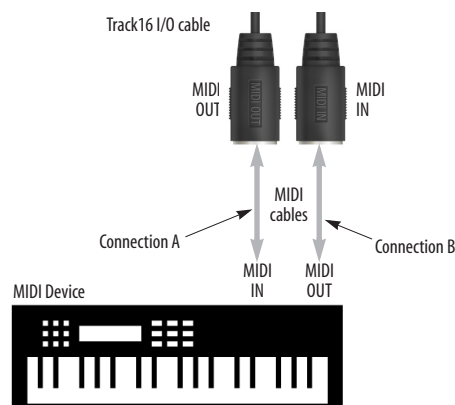


Figure 4-4: Connecting a MIDI device to the Track16.

### One-way MIDI connections

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

### Connecting additional gear with MIDI THRU

If you need to connect several pieces of MIDI gear, run a MIDI cable from the MIDI THRU of a device already connected to the Track16 to the MIDI IN on the additional device as shown below in Figure 4-5. The two devices then share the Track16’s MIDI OUT port. This means that they share the same set of 16 MIDI channels, too, so try

to do this with devices that receive on only one MIDI channel (such as effects modules) so their receive channels don’t conflict with one another.

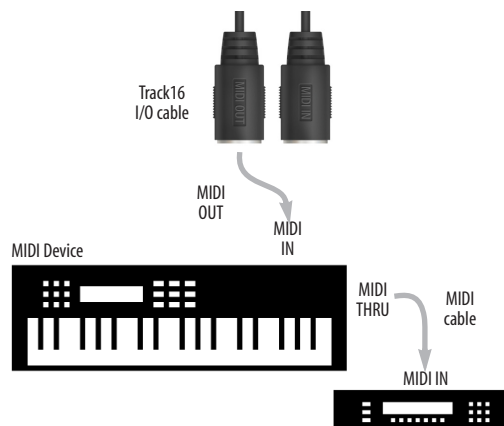


Figure 4-5: Connecting additional devices with MIDI THRU ports.

## POWER OPTIONS

The Track16 can draw power from two possible sources:

1. a computer connected by FireWire, or
2. a DC power supply.

If you use USB to connect to your computer, you must also use the DC power supply included with your Track16.

### Bus power requirements

The Track16 draws all the power it needs from the FireWire bus connection to the computer. However, the FireWire connection to the computer must meet all of the requirements discussed below.

#### 9-pin and 6-pin FireWire connectors

The Track16 can only draw power over the FireWire bus from a 9-pin to 9-pin cable, or a 6-pin to 9-pin cable. It cannot draw power from a FireWire cable with a 4-pin connector, as shown below:

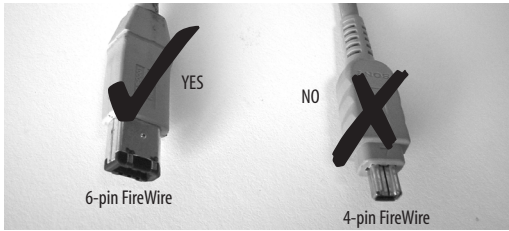


Figure 4-6: 4-pin FireWire connectors cannot be used for bus power.

### When operating under bus power, daisy-chaining is not recommended

The Track16 can be daisy-chained with other FireWire devices from a single FireWire connection to the computer. However, if the Track16 is operating under bus power, this is not recommended. If you need to daisy chain the Track16 with other devices on the same FireWire bus, power the Track16 with the included power adapter. The other devices on the chain should also have their own power supply. In general, bus-powered FireWire devices should not be daisy-chained.

### FireWire adapter products must be powered

If you are using a FireWire adapter (a third-party product that supplies one or more FireWire ports to your computer), it must have direct access to a power supply:

- **PCI FireWire cards** — If you plan to connect the Track16 to a PCI card and run the Track16 under bus power, the PCI card must have a direct connection to the power supply harness inside your computer. This is the same power supply harness to which you connect internal hard drives, CD/DVD drives, etc.
- **PCMCIA or ExpressCard slot adapters** — If you plan to connect the Track16 to a card adapter (inserted in the card slot in your laptop), it must provide a 6-pin or 9-pin connection and it must also have its own power supply. Most commonly, these types of products have a DC power adapter.

As you can see, however, this situation does not allow for remote battery operation, as the PC card adapter requires AC.

### Examples of bus-powered operation

Here are a few typical examples of bus-powered Track16 operation:

#### Bus power from a desktop computer

Your desktop computer is running off of its usual AC power connection, and the Track16 draws power from the FireWire cable connected to the computer. There are no limits to running time.

#### Bus power from an AC-powered laptop

This scenario is identical to the desktop situation described above: the laptop is powered by AC, the Track16 is powered via the FireWire bus and there are no limits to running time.

#### Bus power from a battery-powered laptop

The laptop is being powered by its own battery, and the Track16 is being powered by its FireWire connection to the computer. So the laptop battery is supplying power to both the laptop and the Track16. This is the most compact and portable operating scenario. Running time is determined by the capacity of the laptop battery. For extended recording sessions, bring extra, fully charged laptop batteries.

### DC power supply

If you do not want the Track16 to draw power from the computer, and AC power is available, you can power the Track16 from any standard 9-18 volt, 5 watt DC power supply with any polarity (tip positive or negative), and amperage as shown below.

Voltage	Amperage
9 volts	1.33 amps
12 volts	1 amp
18 volts	0.66 amps

## A TYPICAL TRACK16 SETUP

Here is a typical Track16 studio setup. This rig can be operated without a conventional mixer. All mixing and processing can be done either in the Track16, in the computer with audio software, or both. During recording, you can use the Track16's

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

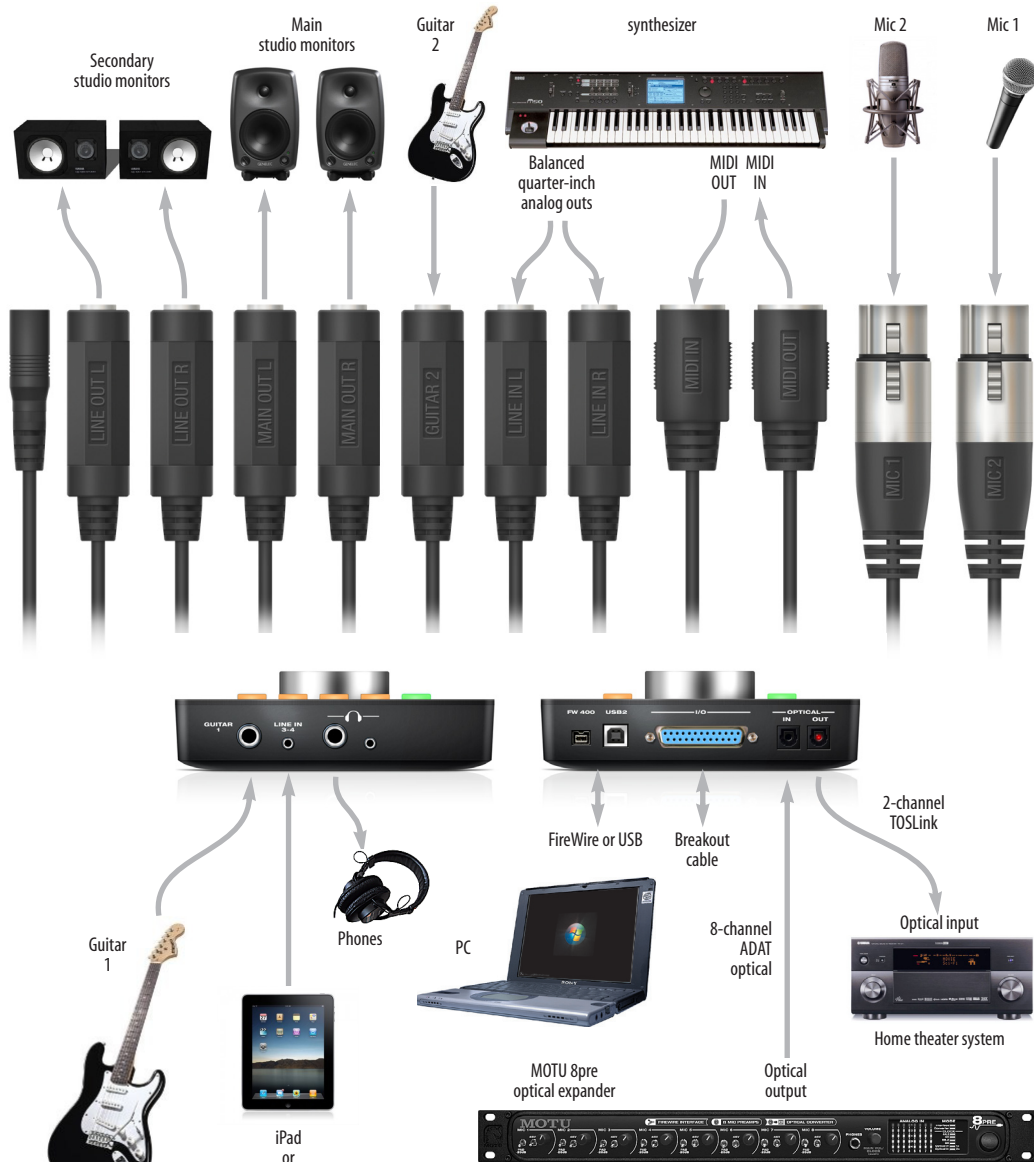


Figure 4-7: A typical Track16 studio setup.

## CONNECTING MULTIPLE MOTU FIREWIRE INTERFACES

You can daisy-chain up to three MOTU FireWire interfaces on a single FireWire bus, but do not run the Track16 or other bus-powered interfaces under bus power when doing so.

When connecting multiple MOTU interfaces, MOTU Audio Console displays the settings for one interface at a time. To view the settings for an interface, click its tab.

### Synchronizing multiple interfaces

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

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

### Operating multiple FireWire interfaces at high sample rates

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



*Part 2*

*Using Track 16*





# CHAPTER 5 MOTU Audio Console

## OVERVIEW

MOTU Audio Console gives you access to basic Track16 hardware settings, such as sample rate, clock source, optical format and more.

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## ACCESSING THE TRACK16 SETTINGS

There are several ways to access Track16 settings:

- Open MOTU Audio Console from the Windows Start menu > Programs > MOTU.
- From within your host audio software's audio driver preferences. For example, in Cubase or Nuendo, open the Device Setup window, click *VST Audio System* and choose *MOTU Audio ASIO* from the *ASIO Driver* menu. Then click the *MOTU Audio ASIO* item in the list and click the *Control Panel* button.

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

### General tab settings

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

### Track16 tab settings

The Track16 tab (Figure 5-1) provides settings that apply to a specific Track16 interface. If you have several Track16 (or other MOTU) interfaces connected, you'll see a separate tab for each one.

## 'GENERAL' TAB SETTINGS

### Sample Rate

Choose the desired *Sample Rate* for recording and playback. The Track16 can operate at 44.1 (the standard rate for compact disc audio), 48, 88.2, 96, 176.4 or 192KHz. If you are operating at a sample rate between 44.1 and 96kHz, make absolutely sure that all of the devices connected digitally to the Track16's optical connectors match the Track16's sample rate. At the 4x sample rates (176.4 or 192kHz), optical I/O is disabled.

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

### Master Clock Source

The *Master Clock Source* determines the digital audio clock that the Track16 will use as its time base. The following sections briefly discuss each clock source setting.

#### Internal

Use the *Internal* setting when you want the Track16 to operate under its own digital audio clock. For example, you may be in a situation where all you are doing is playing tracks off hard disk in your digital audio software on the computer. In a situation like this, you most often don't need to reference an external clock of any kind.

#### ADAT optical

The *ADAT optical* clock source setting refers to the clock provided by the Track16's optical input, when it is connected to another optical device. This setting only appears in the Master Clock Source

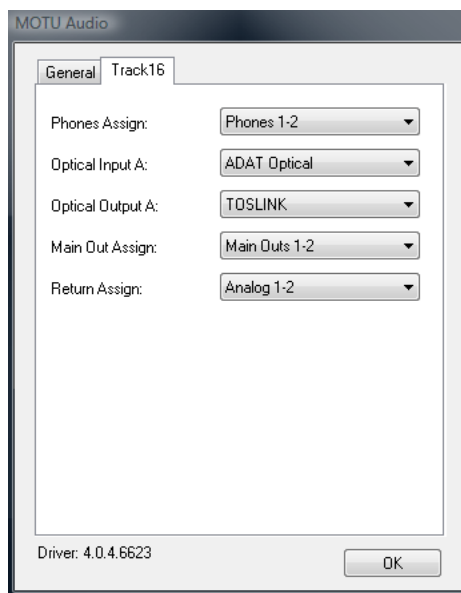
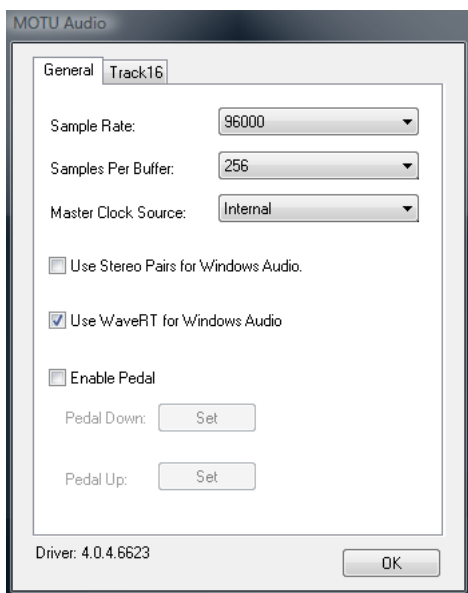


Figure 5-1: MOTU Audio Console gives you access to all of the settings in the Track16 hardware.

menu when the optical input bank is enabled and set to the *ADAT Optical* format, as explained in “Optical input/output” on page 36.

This setting can be used to resolve the Track16 directly to the optical input connection, allowing you to resolve the Track16 to the other device.

This setting is useful if you just need to make a simple, click-free digital transfer between the Track16 and another device.

For further details about this setting, see “Choosing a clock source for optical connections” on page 25.

#### TOSLINK

The *TOSLINK* clock source setting refers to the clock provided by the Track16’s optical input, when it is connected to another optical device. This setting only appears in the Master Clock Source menu when the optical input bank is enabled and set to the *TOSLINK* format, as explained in “Optical input/output” on page 36.

This setting can be used to resolve the Track16 directly to the optical input connection, allowing you to resolve the Track16 to the other device.

This setting is useful if you just need to make a simple, click-free digital transfer between the Track16 and another device.

For further details, see “Choosing a clock source for optical connections” on page 25.

#### SMPTE

Choose this setting to resolve the Track16 directly to SMPTE time code (LTC) being received via one of the Track16’s audio inputs. For details, see “Setup for SMPTE time code sync” on page 104 and chapter 10, “MOTU SMPTE Console” (page 101).

#### Samples Per Buffer

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

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

☛ If you don’t need to process an incoming live signal with software plug-ins, you can monitor the signal with no delay at all using CueMix FX, which routes the signal directly to your speakers via hardware. For details, see chapter 9, “CueMix FX” (page 55).

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

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

This setting presents you with a trade-off between the processing power of your computer and the delay of live audio as it is being processed by plug-ins. If you reduce the *Samples Per Buffer*, you reduce patch thru latency, but significantly increase the overall processing load on your computer, leaving less CPU bandwidth for things like real-time effects processing. On the other hand, if you increase the *Samples Per Buffer*, you reduce the load on your computer, freeing up bandwidth for

effects, mixing and other real-time operations. But don't set the *Samples Per Buffer* too low, or it may cause distortion in your audio.

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

If you are at a point in your recording project where you are not currently working with live, patched-thru material (e.g. you're not recording vocals), or if you have a way of externally monitoring input, choose a higher *Samples Per Buffer* setting.

Depending on your computer's CPU speed, you might find that settings in the middle work best.

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

#### **Monitoring live inputs without plug-in effects**

As mentioned earlier, CueMix FX allows you to monitor dry, unprocessed live inputs with no delay at all. For complete details, see chapter 8, "Reducing Monitoring Latency" (page 49).

#### **Use stereo pairs for Windows audio**

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

#### **Use WaveRT for Windows audio**

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

#### **Enable Pedal**

This setting applies to other MOTU audio interfaces, but it does not apply to the Track16.

### **TRACK16 TAB SETTINGS**

#### **Phones Assign**

The *Phones Assign* setting lets you choose what you will hear from the headphone jack. Choose *Main Out 1-2* if you'd like the headphone output to match the main outs. Choose *Phones 1-2* if you would like the headphones to serve as their own independent output, which you can access as an independent output destination in your host audio software and as an output destination for the four on-board CueMix FX mix busses.

#### **Optical input/output**

The *Optical input* and *Optical output* settings let you choose between 8-channel ADAT optical ('lightpipe') and stereo S/PDIF optical ('TOSLINK') as the format for the Track16's optical input and output. Choose the format that matches the device connected. If you are not using the optical connections, it is recommended that you turn them off (by choosing *Disabled* from the menu) to reduce bandwidth and processing overhead. Note that you can operate each optical port independently. For example, you could use the ADAT optical format on the input (for a digital mixer) and optical S/PDIF on the output (for a DAT deck).

#### **Main Out Assign**

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

## Return Assign

The *Return Assign* menu lets you choose any pair of Track16 audio outputs. The audio signal from this output pair is then sent back to the computer via the *Stereo Return 1-2* bus. This stereo return bus from the Track16 appears in your host software alongside all other Track16 inputs, wherever your host software lists them.

The Track16 stereo return bus can be used for a variety of purposes. For example, you could use it to send a final mix being played through the Track16 back to the computer, where you could record it for mastering or archiving purposes.

As another example, you could use the stereo return bus to capture tracks played from your host software, along with live inputs being routed directly through the Track16 hardware via CueMix FX (with or without CueMix effects processing on the live inputs).



## CHAPTER 6 Hardware operation

### METERS

The first time you switch on the Track16, its LED meters show signals indicated by the white labels below them. Push the METERS button to toggle the meters to display the signals indicated by the blue labels. The METERS button glows blue. Push it again to return to the white labels (dim blue).

### KNOB

At any time, the knob controls one of three things:

1. output volume
2. input trim
3. input bus level

The buttons let you quickly choose what the knob does.

### BUTTONS

The first time you switch on the Track16, the state of the buttons is as follows:

Button	LED color
METERS	Off
POWER	Blue
Currently selected input or output	Green
Other inputs and outputs	Amber
Phantom power (V) and Pad (P) LEDs	Off

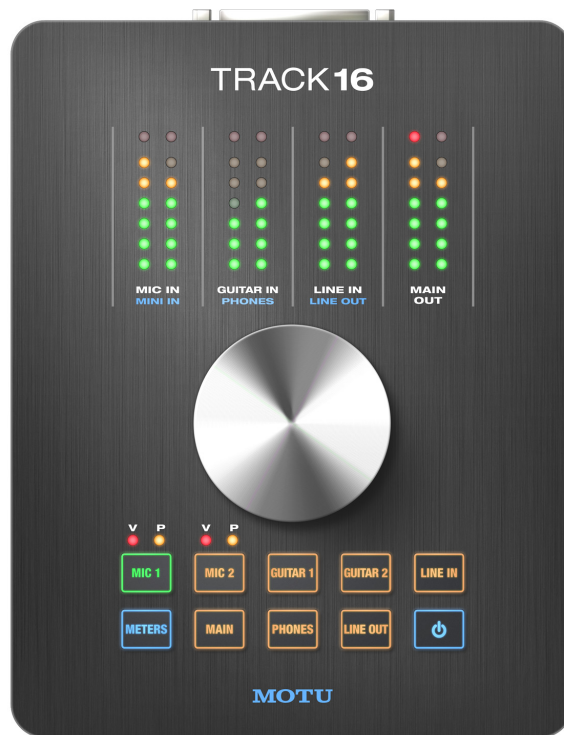


Figure 6-1: Use the knob and buttons to control basic Track16 functions.

### Output volume control

To control volume, push the desired output (MAIN, PHONES, or LINE OUT). The button illuminates green. Turn the knob to adjust the output volume.

### Input trim

To control input trim, push the desired input (MIC, GUITAR, or LINE IN). The button illuminates green. Turn the knob to adjust the input trim.

### Input bus volume

To control input bus level (CueMix input fader), push the output bus (MAIN, PHONES, LINE OUT) and the desired input (MIC, GUITAR, or LINE IN) at the same time. Both the input and output buttons illuminate green. The input button illuminates green. Turn the knob to adjust the bus volume.

### Muting/unmuting

To mute or unmute an output, push the desired output (MAIN, PHONES, or LINE OUT) to select it. The button illuminates green. Push the knob to mute or unmute the output. While muted, the button blinks.

To mute or unmute input bus volume (CueMix input fader), push the output bus (MAIN, PHONES, LINE OUT) and the desired input (MIC, GUITAR, or LINE IN) at the same time. Both the input and output buttons illuminate green. Push the knob once to mute or unmute the input on that bus. When muted, the button blinks; unmuted it stops blinking.

### Pad and 48V phantom power

To toggle a -20dB pad, push MIC 1 or MIC 2 and then push the knob once. The yellow pad (P) LED turns on or off.

To toggle 48V phantom power, push MIC 1 or MIC 2 and hold down the knob. The red phantom power (V) LED turns on or off.

### Lockout mode

Hold the METERS button down for 2 seconds to enter Lockout mode. In this mode, the knob and buttons are temporarily disabled, to prevent unwanted changes to any Track16 settings. While in lockout mode, the METERS button blinks.

Hold the METERS button for 2 seconds again to exit lockout mode.

### Button LED color schemes

The Track16 button LEDs have the ability to display several different color combinations. From the factory, the currently selected button is green and the others are amber. But you can choose other color combinations, if you like.

To change the button LED color scheme:

- 1 Hold the METERS button down for 2 seconds to enter Lockout mode.

The METERS button blinks.

- 2 Press the METERS button repeatedly to cycle among the different LED color schemes.

- 3 When you have chosen the desired scheme, hold the METERS button again for 2 seconds to exit Lockout mode.

You can change the LED color scheme at any time.

### FACTORY SETTINGS

To restore these factory settings, hold down the MAIN and LINE OUT buttons for 2 seconds:

- The MAIN button is active (green) and, therefore, the knob controls main out volume.
- The OUTPUT METERS button is inactive, and therefore, the meters display levels for MIC IN, GUITAR IN, and LINE IN.
- On the mic inputs, 48V phantom power and -20 dB pad are off.



# CHAPTER 7 **Configuring Host Audio Software**

## OVERVIEW

The Track16 provides multi-channel audio input and output for ASIO- and WDM-compatible audio applications, including Ableton Live, Avid Pro Tools, Cockos Reaper, Propellerhead Reason and Record, Steinberg Cubase and Nuendo, Cakewalk SONAR, and others.

Additionally, sample-accurate synchronization is possible with applications that support ASIO 2 or 3's sample-accurate positioning protocol.

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## PREPARATION

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

- chapter 3, "Installing the Track16 Software" (page 19)
- chapter 4, "Installing the Track16 Hardware" (page 21).

## RUN MOTU AUDIO CONSOLE

Before you run your host audio software, launch MOTU Audio Console to configure your Track16 hardware.

MOTU Audio Console provides settings to configure your audio interface and enable the desired inputs and outputs. Only enabled inputs

and outputs will be available to your software, so this is an important step. For complete details see chapter 5, "MOTU Audio Console" (page 33).

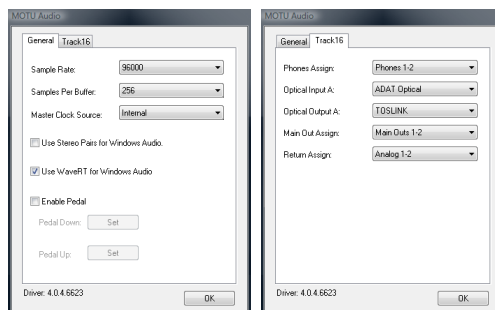


Figure 7-1: MOTU Audio Console.

For complete details about the Track16 settings, see chapter 5, "MOTU Audio Console" (page 33). The following sections provide a brief explanation of each Track16 setting.

### Sample rate

Choose the desired overall sample rate for the Track16 system and your host audio software. Newly recorded audio will have this sample rate.

### Master Clock Source

This setting is very important because it determines which audio clock the Track16 will follow.

If you do not have any digital audio connections to your Track16 (you are using the analog inputs and outputs only), and you will not be slaving your host software to external SMPTE time code, choose *Internal*.

If you have devices connected to the Track16 optical ports, see "Choosing a clock source for optical connections" on page 25.

If you are slaving the Track16 and your host software to SMPTE time code, follow the directions in chapter 10, “MOTU SMPTE Console” (page 101).

☛ In order to use the Track16’s direct SMPTE sync (and sample-accurate sync) feature, the host audio software must support the ASIO 2.0 sample-accurate positioning protocol. Cubase and Nuendo support this protocol; for other software, check with its documentation.

### Samples Per Buffer

The *Samples Per Buffer* setting can be used to reduce the delay — or *monitoring latency* — that you hear when live audio is patched through your Track16 hardware and host audio software. For example, you might have MIDI instruments, samplers, microphones, and so on connected to the analog inputs of the Track16. If so, you will often be mixing their live input with audio material recorded in your audio software. See chapter 8, “Reducing Monitoring Latency” (page 49) for complete details.

### Phones Assign

This Track16 setting lets you choose what you’ll hear from the headphone jack. For example, if you choose *Main Out 1-2*, the headphones will duplicate the main outs. Or you can choose any other output pair. If you choose *Phones 1-2*, this setting makes the headphone jack serve as its own independent output pair. As a result, you’ll see *Phones 1-2* as an additional audio destination in your host’s audio output menus.

☛ The 1/8th-inch “mini” headphone jack always mirrors the 1/4-inch jack.

### Optical input and output

To make a Track16 optical input or output available in your host software, choose the appropriate format (*ADAT optical* or *TOSLINK*) from the optical input and/or output menu. If you won’t be using the optical connectors, turn them off.

### Main Outs Assign

Use the Main Outs Assign setting to determine what audio you will hear on the main outs of the Track16. If you would like to treat them as their own separate output pair, choose *Main Out 1-2*.

### Return Assign

In your host audio software audio input menus, you’ll see an Track16 input called *Return 1-2*. This is a stereo feed from the Track16 that matches the signal of one of its output pairs. Use the Return Assign menu in MOTU Audio Console to choose which output pair you would like to hear on this return. This can be used, for example, to record back a final stereo mix that includes effects processing from the Track16 DSP (such as the Leveler) for reference and archiving purposes.

☛ Warning: the Return inputs can cause feedback loops! **DO NOT** assign this input to a track that shares the same Track16 output pair as the returns.

### Reverb return

The Track16 also supplies a return to your host software that carries the output of its reverb processor. This return can be used for any purpose you wish.

## CHOOSING THE MOTU AUDIO DRIVER

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

## Pro Tools

In Avid Pro Tools, go to the Setup menu and choose *Playback Engine* as shown in Figure 7-2. Choose the *MOTU Audio ASIO* from the Current Engine menu. For information about the *H/W Buffer Size* setting, see "Adjusting the audio I/O buffer" on page 51.

👉 Pro Tools 9 or later is required for operation with the Track16.

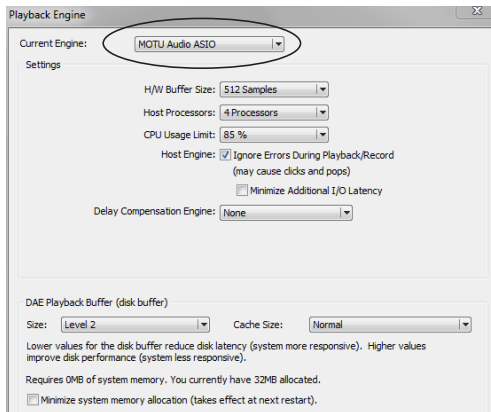


Figure 7-2: Enabling the MOTU Audio ASIO driver in Pro Tools

## Cubase and Nuendo

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

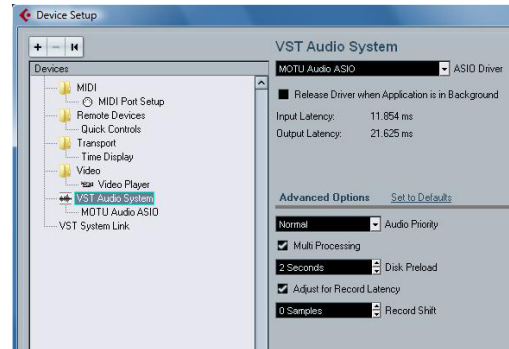


Figure 7-3: Enabling the MOTU Audio ASIO driver in Cubase

## Live

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

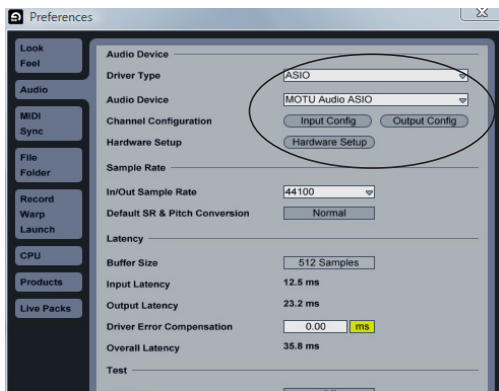


Figure 7-4: Enabling the MOTU Audio ASIO driver in Live

## Reason and Record

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

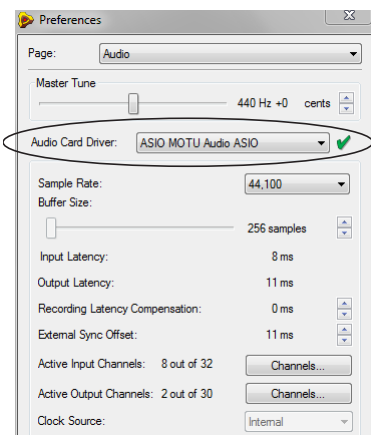


Figure 7-5: Enabling the MOTU Audio ASIO driver in Reason

## Reaper

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

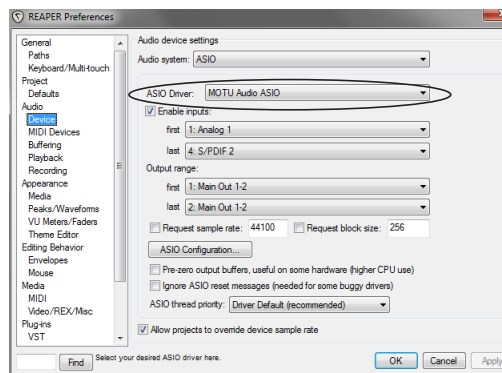


Figure 7-6: Enabling the MOTU Audio ASIO driver in Reaper

## SONAR

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

### Using the MOTU ASIO driver

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

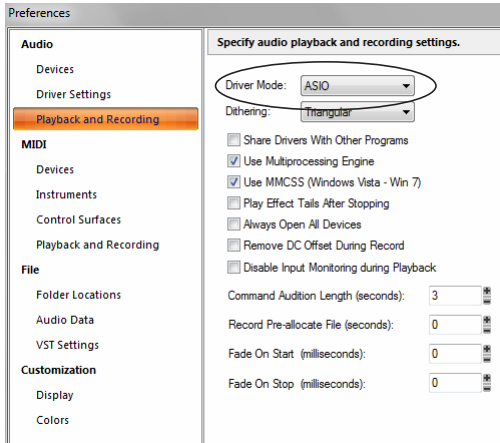


Figure 7-7: Enabling the MOTU Audio ASIO driver in SONAR.

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

**5** Check the Track16 inputs and outputs that you wish to use and uncheck the ones you don't.

### Using the MOTU WDM driver

**1** Go to SONAR's Preferences.

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

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

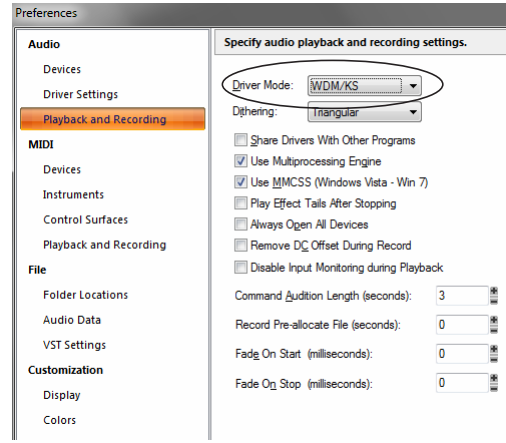


Figure 7-8: Enabling the MOTU WDM driver in SONAR.

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

**5** Check the Track16 inputs and outputs that you wish to use and uncheck the ones you don't, as shown in Figure 7-7 on page 45.

### Other audio settings in SONAR

There are a few other settings in SONAR that impact Track16 operation:

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

**2** Choose an Track16 input and output for recording and playback timing as shown below in Figure 7-9. It doesn't matter which input or output you choose.

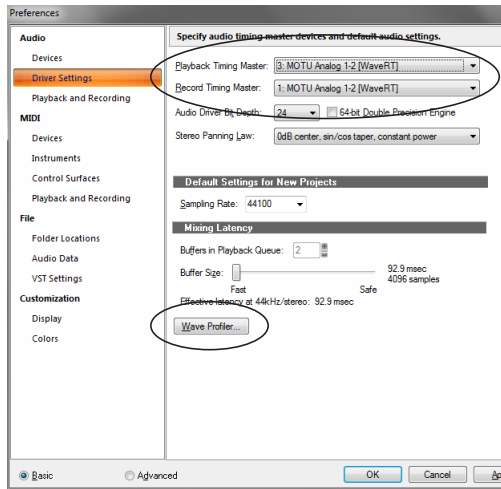


Figure 7-9: Make sure you have chosen a Track16 input and output for the playback and recording timing master settings.

3 Additionally, if you are using the MOTU WDM driver, click the *Wave Profiler* button and run the Wave Profiler. This process chooses the optimum settings for the Track16 hardware.

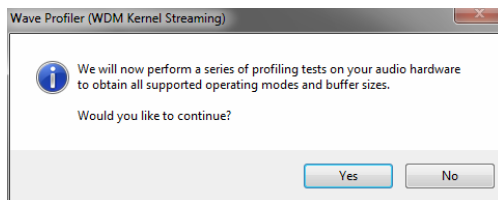


Figure 7-10: Running the Wave Profiler in Cakewalk's SONAR.

## Sound Forge

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

## Other audio software

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

## REDUCING LATENCY

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

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

For information, see the "Adjusting the audio I/O buffer" section of chapter 8, "Reducing Monitoring Latency" (page 49).

## WORKING WITH TRACK16 INPUTS AND OUTPUTS

Once you've enabled the Track16's driver, Track16 audio inputs and outputs will appear in your host software wherever audio inputs and outputs are listed. If you don't see the optical inputs and/or outputs, check MOTU Audio Console to make sure they are enabled. If you don't plan to use either optical bank, turn it off to conserve computer bandwidth.

Remember that the main outs and headphone output can be configured in MOTU Audio Console to mirror other outputs. In addition, the Track16 optical banks can be disabled or configured as either ADAT optical or TOSLink. This will affect how they appear (and if they appear at all) in your host software's audio input lists. See "Optical input/output" on page 36, "Phones Assign" on page 36 and "Main Out Assign" on page 36.

## Number of channels

If your host audio software requires that you specify the number of audio voices or channels you will be using, be sure to choose enough channels to cover the 16 inputs and 16 outputs provided by your Track16.

### Mix1 1-2 return bus

In your host audio software audio input menus, you'll see an Track16 input called *Mix1 Return 1-2*. This is a stereo feed from the Track16 that matches its main outs (Mix 1). This can be used, for example, to record a final stereo mix for reference and archiving purposes.

⚠ Warning: the Mix1 Return 1-2 return inputs can cause feedback loops! DO NOT assign this input to a track assigned to the Track16 main outs.

For Mix1 Return 1-2, you can choose whether or not to include audio being sent to the main outs from the computer. See “Mix1 Return Includes Computer” on page 79.

### The “Phones 1-2” output

Host audio tracks assigned to the *Phones 1-2* output pair will be heard on the headphone jack only. If the Phones are set to mirror another output (such as the main outs), the mirrored audio stream is mixed with the phones audio output. For further explanation, see “Phones Assign” on page 36.

### Audio Input and output names

The Track16 audio driver supplies text string labels for its inputs and outputs to clearly identify each one, but some applications do not display these labels. For information on how you can identify each input and output in a numbered list like this, see Appendix A, “Audio I/O reference” page (107).

### PROCESSING LIVE INPUTS WITH HOST PLUG-INS

If you patch a live input (such as a MIDI synthesizer) through a plug-in effect in your host software, you might hear a slight delay. There are several ways to reduce this delay. For details, see chapter 8, “Reducing Monitoring Latency” (page 49).

### WORKING WITH CUEMIX FX MIXING AND EFFECTS

The Track16 provides powerful external mixing, EQ, compression and reverb, which you can operate hand-in-hand with your host's mixing environment. For example, the Track16 can serve as a monitor mixer routing channels to musicians, or it can serve as an integrated extension of your host's mixing environment. If you program an Track16 mixing and processing configuration that goes hand in hand with your host project, be sure to use the file save features in CueMix FX to save the Track16 settings as a file in your host project folder for instant recall of all settings. See chapter 9, “CueMix FX” (page 55) for complete details.

### SYNCHRONIZATION

You may encounter situations in which you need to synchronize your audio software and the Track16 to the other components of your system.

#### Synchronizing digital audio connections

If you have devices connected to the Track16 digital optical ports, you need to be concerned with the synchronization of the Track16's digital audio clock with other devices connected to it digitally (if any). For example, if you have a digital mixer connected to the Track16 via an ADAT optical light pipe cable, you need to make sure that their audio clocks are phase-locked. For details, see “Choosing a clock source for optical connections” on page 25. If you don't have any digital audio devices connected to the Track16, digital audio phase-lock does not apply to you.

#### Resolving directly to time code (with no synchronizer)

If your host audio software supports ASIO2 or 3's sample-accurate positioning protocol, then it can resolve to the Track16's built-in time code synchronization feature. To resolve your Track16 directly to SMPTE time code with no additional synchronization devices, use the setup shown in “Setup for SMPTE time code sync” on page 104.





## CHAPTER 8 Reducing Monitoring Latency

### OVERVIEW

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

This delay is caused by the amount of time it takes for audio to make the entire round trip through your computer, from when it first enters an Track16 input, passes through the Track16 hardware into the computer, through your host audio software, and then back out to an Track16 output.

If you don't need to process a live input with plug-ins, the easiest way to avoid monitoring latency is to use the Track16's CueMix FX digital mixer to patch the input directly to your monitor outs via the Track16 audio hardware. The Track16 even provides effects processing (EQ, compression and reverb), which can be applied on input, output, or even at the bussing stage, just like a conventional mixer. For details, see "CueMix FX hardware monitoring" on page 52.

If you *do* need to process a live input with host software plug-ins, or if you are playing virtual instruments live through your Track16 audio hardware, you can significantly reduce latency — and even make it completely inaudible, regardless of what host audio application software you use. This chapter explains how.

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

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## MONITORING LIVE INPUT

There are two ways to monitor live audio input with an Track16: 1) through the computer or 2) via the Track16 CueMix FX hardware mixer.

Figure 8-1 shows method 1, which allows you to apply host-based effects processing via plug-ins in your audio software. See the next section, “Adjusting the audio I/O buffer” for details about how to reduce — and possibly eliminate — the audible monitoring delay that the computer introduces.

Figure 8-2 shows how to use CueMix FX hardware-based monitoring, which lets you hear what you are recording with no monitoring delay and no computer-based effects processing. Instead, input

is routed directly to an output, either with or without Track16-based effects processing (EQ, compression or reverb). See “CueMix FX hardware monitoring” later in this chapter for details on how to use CueMix FX with your audio software, or with the included CueMix FX software.

If the material you are recording is suitable, there is a third way to monitor live input: use both methods (Figure 8-1 and Figure 8-2) at the same time. For example, you could route guitar to both the computer (for an amp model effect) and mix that processed signal on the main outs with dry guitar from CueMix FX — or perhaps with a touch of Track16 Classic Reverb.

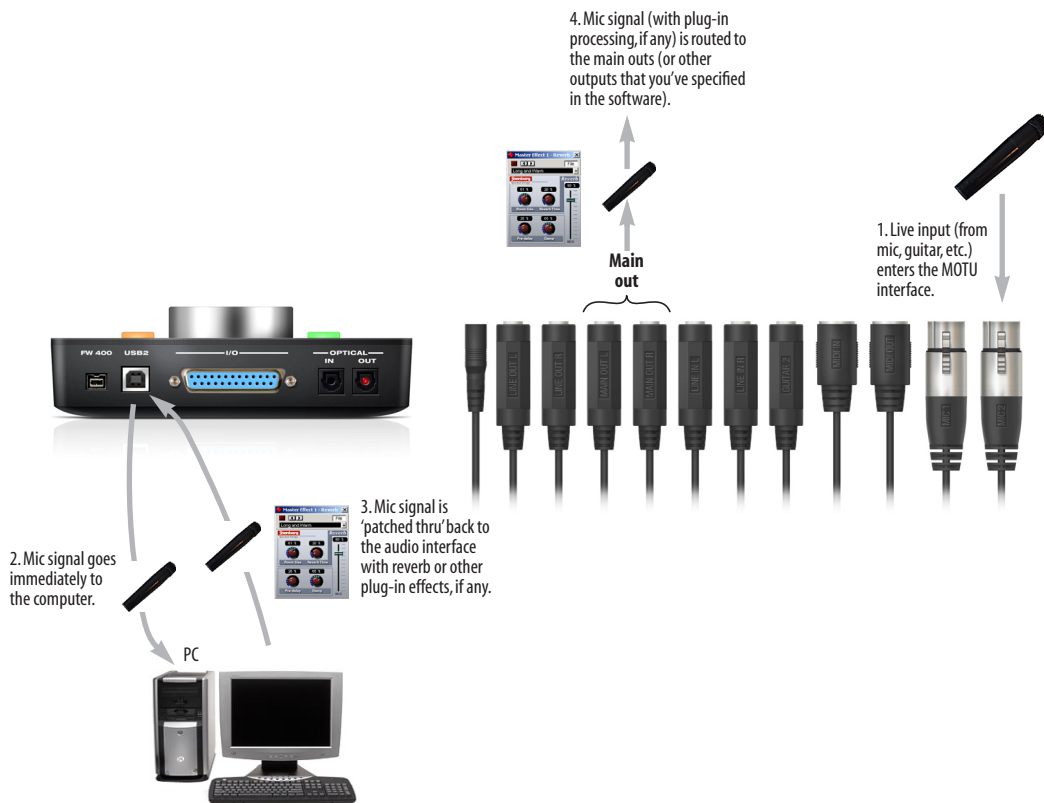


Figure 8-1: There are two ways to monitor live audio inputs with an Track16: 1) through the computer or 2) via CueMix FX hardware monitoring. This diagram shows method 1 (through the computer). When using this method, use your host software's buffer setting to reduce the slight delay you hear when monitoring the live input, but don't lower it too much, or your computer might get sluggish.

## ADJUSTING THE AUDIO I/O BUFFER

A *buffer* is a small amount of computer memory used to hold data. For audio interfaces like the Track16, buffers are used for the process of transferring audio data in and out of the computer. The size of the buffers determines how much delay you hear when monitoring live inputs through your audio software: larger buffers produce more delay; smaller buffers produce less.

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

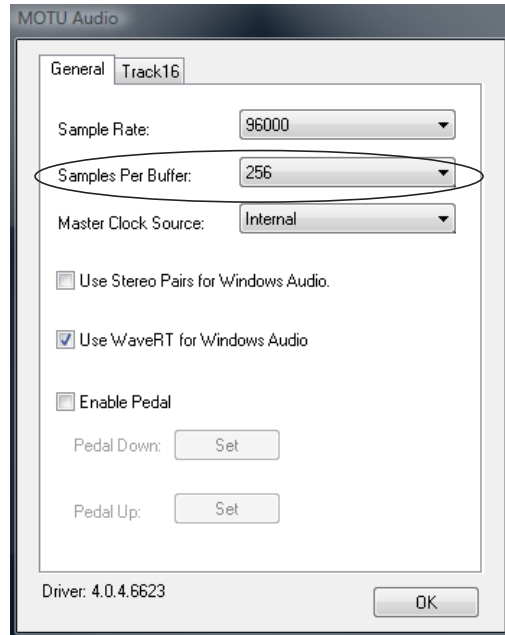


Figure 8-3: Lowering the 'Samples Per Buffer' setting in MOTU Audio Console reduces patch thru latency. But doing so increases the processing load on your computer, so keep an eye on the Performance Monitor in your host audio software.

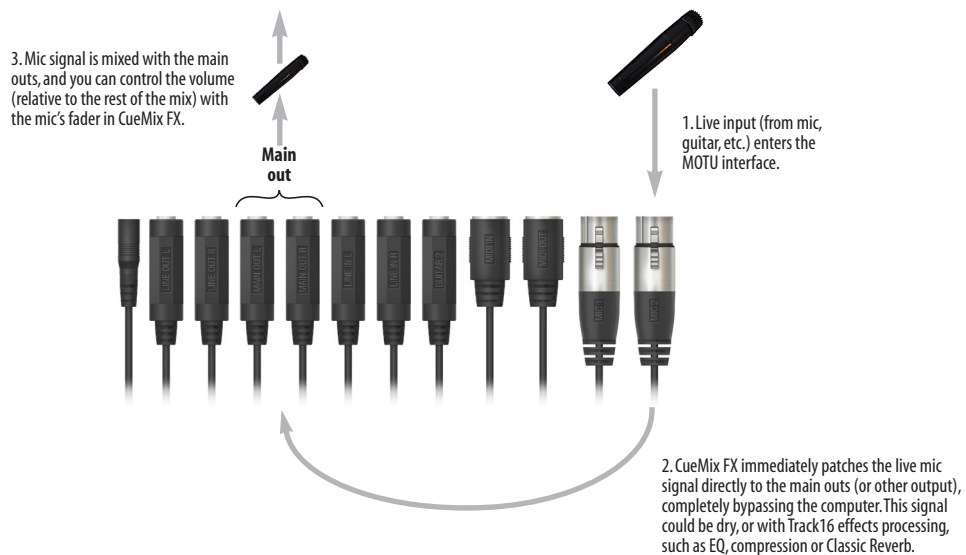


Figure 8-2: This diagram shows the signal flow when using CueMix FX no-latency monitoring. Notice that this method does not process the live input with plug-ins in your audio software. Instead, you can apply Track16 effects, such as the reverb, EQ and/or compression.

### Lower latency versus higher CPU overhead

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

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

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

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

### Transport responsiveness

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

### Effects processing and automated mixing

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

### CUEMIX FX HARDWARE MONITORING

The Track16 has a more direct method of patching audio through the system: *CueMix FX*. CueMix FX is a digital mixer in the Track16 hardware itself. CueMix FX has two important benefits:

- First, it completely eliminates the patch thru delay (reducing it to a small number of samples — the same amount as any digital mixer).
- Secondly, CueMix FX imposes no strain on the computer.

The trade-off, however, is that CueMix FX bypasses your host audio software. Instead, live audio inputs are patched directly through to outputs in the Track16 itself and are mixed with disk tracks playing back from your audio software. This means that you cannot apply host-based plug-ins, mix automation, or other real-time effects that your audio software provides. But for inputs that don't need these types of features, CueMix FX is the way to go.

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

### TWO METHODS FOR CONTROLLING CUEMIX FX

There are two ways to control CueMix FX:

- With the Track16's CueMix FX software
- From within your host audio software (if it supports direct hardware monitoring)

You can even use both methods simultaneously.

### Using the CueMix FX app

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

CueMix FX allows you to create up to eight separate Track16 stereo mixes, or any other desired routing configurations. These routings are independent of your host audio software. For complete details, see chapter 9, “CueMix FX” (page 55).

### Controlling CueMix FX from your audio software

Some ASIO-compatible audio applications, such as Cubase and Nuendo, allow you to control CueMix FX monitoring from within the application (without the need to use CueMix FX). In most cases, this support consists of patching an Track16 input directly to an output when you record-arm a track. Exactly how this is handled depends on the application.

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

### Controlling CueMix FX from within Cubase or Nuendo

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

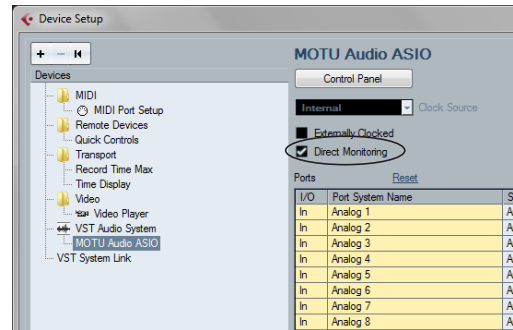


Figure 8-4: Activating the Direct Monitoring option in Cubase or Nuendo.

### Other ASIO 2.0-compatible host software

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

### Using CueMix FX with WDM- or Wave-compatible software

Run CueMix Console (chapter 9, “CueMix FX” (page 55)) and use it to route live inputs directly to outputs, control their volume and panning, etc. If necessary, you can save your CueMix Console setup with your project file.



## CHAPTER 9 CueMix FX

### OVERVIEW

CueMix FX is a cross-platform software application that provides graphic, on-screen control for the Track16's flexible CueMix FX on-board mixer and effects processing.

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

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

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### A 16-BUS MIXER WITH EQ, COMPRESSION AND REVERB

All Track16 inputs can be routed to the on-board CueMix FX 16-bus (8 stereo bus) digital mixer driven by hardware-based DSP with 32-bit floating point precision.

The CueMix FX mixer allows you to apply no-latency effects processing to inputs, outputs or busses directly in the Track16 hardware, independent of the computer. Effects can even be applied when the Track16 is operating stand-alone (without a computer). Input signals to the computer can be recorded wet, dry, or dry with a wet monitor mix (for musicians during recording, for example). Effects include:

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

The Track16's flexible effects architecture allows you to apply EQ and compression on every input and output (a total of 32 channels), with enough DSP resources for at least one band of parametric EQ and compression on every channel at 48 kHz. DSP resources are allocated dynamically and a DSP meter in the CueMix FX software allows you to keep tabs on the Track16's processing resources.

Each input, output and mix bus provides a send to the Classic Reverb processor, which then feeds reverb returns to mix busses and outputs.

## ADVANTAGES OVER HOST-BASED MIXING AND PROCESSING

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

- CueMix has no buffer latency. Thanks to the Track16's DSP chip, CueMix provides the same throughput performance as a digital mixer.
- CueMix mixing and effects processing imposes no processor drain on the computer's CPU.
- CueMix routing can be maintained independently of individual software applications or projects.
- CueMix routing can operate without the computer, allowing the Track16 to operate as a portable, stand-alone mixer with effects.

## OPENING CUEMIX FX

*CueMix FX* is an application installed with the rest of your Track16 setup software. It can be found in the Start menu.

## CUEMIX FX BASIC OPERATION

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

### A separate mix bus for each output pair

The Track16 provides a separate mix bus for each of its output pairs, as follows:

Mix name	Physical output
Bus 1	Main Out 1-2
Bus 2	Phones 1-2
Bus 3	Analog 1-2
Bus 4	ADAT 1-2, TOSLink 1-2, or disabled. See note below.*
Bus 5	ADAT 3-4 or disabled.*
Bus 6	ADAT 5-6 or disabled.*
Bus 7	ADAT 7-8 or disabled.*
Bus 8	Mix Return 1-2

\*If the Track16's optical output is disabled in MOTU Audio Console (Figure 5-1 on page 34), CueMix busses 4 through 7 are also disabled. If the optical output bank is set to TOSLink, bus 4 is enabled for TOSLink channels 1-2. If the optical output bank is set to ADAT, then busses 4 through 7 are enabled for the four optical bus pairs indicated in the table above. For details about configuring the optical output, see "Optical input/output" on page 36.

### Many inputs to one output pair

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

### Viewing one mix bus at a time

CueMix FX displays one mix bus at a time in the *Mixes* tab (Figure 9-2 on page 58). To select which mix you are viewing, choose it from the mix bus menu (Figure 9-2). The mix name appears above the mix bus master fader (Figure 9-2), where you can click the name to change it.

### Each mix bus is independent

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

### Input channels

The *Inputs* tab (Figure 9-3 on page 60) gives you access to settings for individual Track16 inputs (or input pairs), such as phase, trim, EQ and dynamics processing. Each input also includes a send to the Track16's global reverb processor. These settings are applied to the signal before it goes anywhere else (to a mix bus or the computer).



## Output channels

The *Outputs* tab (Figure 9-6 on page 63) gives you access to settings for each Track16 output pair, including EQ, dynamics processing and send/return controls for feeding and returning the output signal to/from the Track16's global reverb processor. These settings are applied to the signal just before it is sent to the output.

## Channel focus and settings

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

## Global reverb processor

The Track16 has a global reverb module (Figure 9-23 on page 75). Once it has been activated, you can feed signals to the reverb processor from various points in the Track16 mix matrix via input sends, bus sends and output sends. Stereo output from the reverb processor can then be fed back to mix busses or output pairs using reverb returns.

## Other features

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

## Widening the CueMix FX window

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

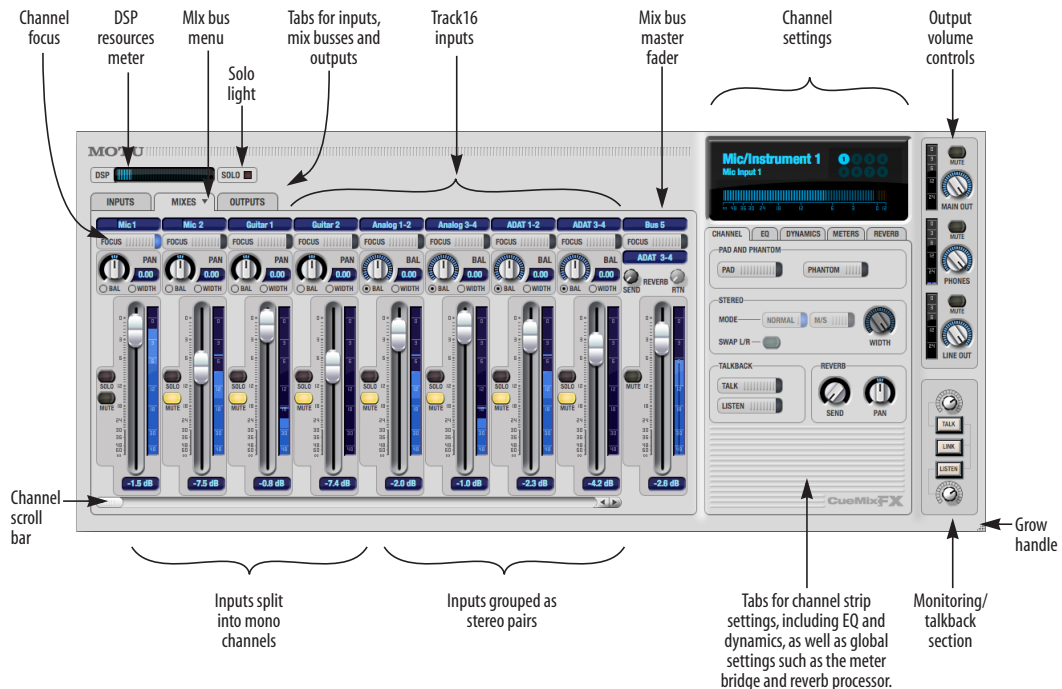


Figure 9-1: CueMix FX is a virtual mixer that gives you control over the Track16's on-board mixing features.

## THE MIXES TAB

Click the *Mixes* tab (Figure 9-2) to gain access to the Track16's eight stereo mix busses. The Mixes tab displays one mix bus at a time.

### Viewing a mix

Choose the mix you wish to view from the mix bus menu (in the Mixes tab itself, as shown in Figure 9-2). The menu shows all mixes by name, followed by the Track16 output pair to which each bus master fader is assigned. If the optical output bank is either disabled or set to TosLink (see "Optical input/output" on page 36), then bus 5, 6 and 7 are disabled.

### Naming a mix

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

### Mix bus output

The *bus output* (Figure 9-2) indicates the Track16 stereo output pair to which the mix bus is routed.

### Bus fader

The bus fader (Figure 9-2) controls the overall level of the mix (its volume on its stereo output). Use the individual input faders to the left to control individual input levels.

### Bus mute

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

### Bus level meter

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

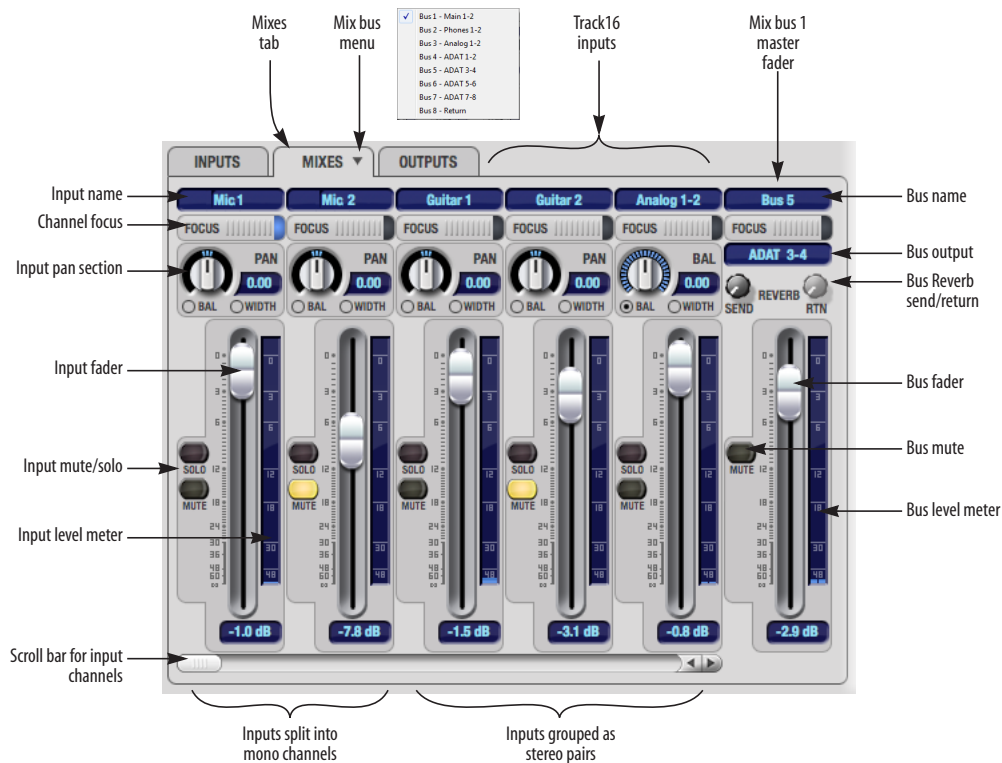


Figure 9-2: The Mixes tab.

### Bus reverb send/return

The bus reverb send (Figure 9-2) feeds the output of the mix bus, pre-fader, to the Track16's global reverb processor, where it is merged with any other signals being fed to the reverb. The reverb's output can then be fed back into the mixer at various return points, including the bus return (discussed below).

The bus reverb *return* (Figure 9-2) feeds the output of the Track16's global reverb processor into the mix bus, pre-fader. This includes any other signals currently being fed to the reverb. The bus reverb return is disabled (grayed out) when the reverb *Split Point* is set to *Output* to eliminate the possibility for feedback loops created by reverb send/return loops. See “Split point” on page 76.

### Input section

The horizontally scrolling area in the Mix tab to the left of the master fader (Figure 9-2) displays channel strips for all currently enabled Track16 inputs.

### Input channel focus

Click the channel focus button (Figure 9-3) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 9-7 on page 64). Clicking the mix bus master fader focus button brings the assigned output into focus, if there is one. Channel focus also determines which channels are being scoped by CueMix's audio analysis tools, as explained in “Choosing channels for audio analysis” on page 80.

### Input pan section

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

### Balance

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

### Width

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

### Input fader and mute/solo

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

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

## THE INPUTS TAB

The Track16 provides many features for managing analog and digital input signals. Some of these features, such as the Track16's digitally controlled analog trims, are implemented in the analog domain; others are implemented in the digital domain as DSP applied to the digital signal (after the A/D converter on analog inputs). Click the *Inputs* tab (Figure 9-3) to access and control all of these input channel settings for each Track16 input or input pair.

### Input tab settings are global

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

record a completely unprocessed input signal, do not apply any changes to it in the Input tab. The only exception to this is the reverb send, which simply splits the input signal and feeds a copy of it to the Track16's reverb processor.

### Signal flows from top to bottom

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

### Input name

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

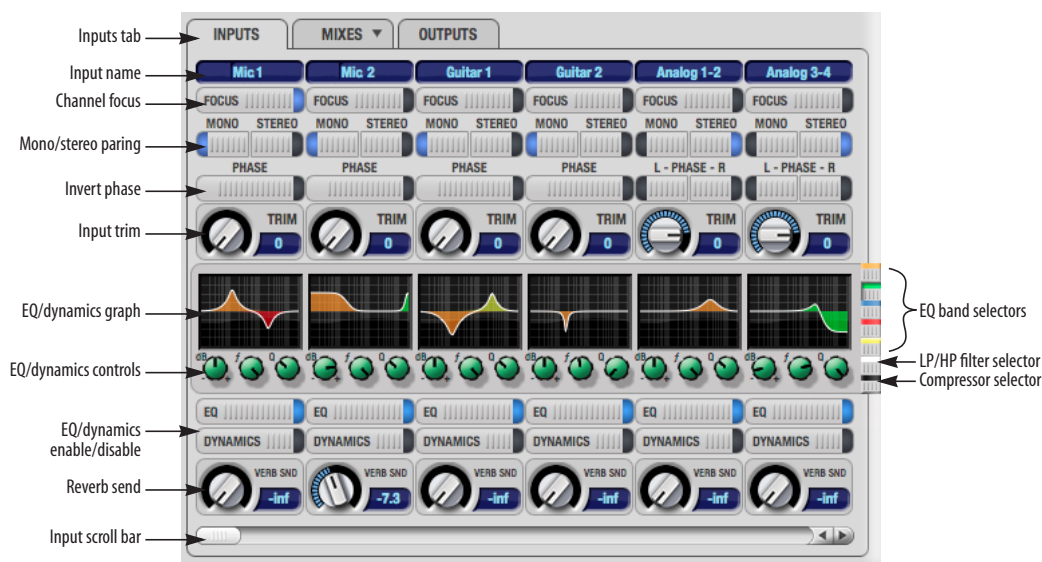


Figure 9-3: The Inputs tab.

### Input channel focus

Click the channel focus button (Figure 9-3) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 9-7 on page 64).

### Mono/stereo pairing

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

### Invert phase

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

### Input trim

All Track16 inputs, both analog and digital, offer continuously variable input trim. In all cases, trim level can be controlled digitally in 1 dB increments. This includes the digitally controlled analog trims on the eight analog inputs. Here is a summary of input trim ranges for each type of Track16 input:

Input	Trim cut	Trim boost	Trim Range
Mic inputs	0 dB	+60 dB	60 dB
Guitar inputs	0 dB	+22 dB	22 dB
Analog inputs	-96 dB	+22 dB	118 dB
ADAT optical	0 dB	+12 dB	12 dB
S/PDIF (TOSLINK)	0 dB	+12 dB	12 dB

Once you adjust the trim levels, you can save them as a file on disk for future instant recall. See “Saving and loading hardware presets” on page 79 and “Configurations menu” on page 97.

### Input EQ and dynamics

The Track16 lets you apply 7-band parametric EQ and dynamics processing (DSP) to any input, analog or digital.

The controls in the EQ/Compression section of the Inputs tab (Figure 9-3) let you edit EQ and compression settings within the context of the channel strip. This is ideal when you are comparing settings among neighboring channels, or perhaps even applying the same setting across all inputs. However, for more detailed editing of EQ and compression settings for an input channel, you can click its Focus button and view the settings in the Channel Section of the CueMix FX window (Figure 9-1). This section even provides graphical editing of EQ curves and the compressor graph, allowing you to click and drag directly on the graphic. For details see “The channel settings section” on page 64.

### The EQ/Dynamics graph

The EQ/Dynamics graph for each input channel strip (Figure 9-3) provides a thumbnail view of the EQ curves or Compressor graph for the channel. This graphic is for display purposes only; it cannot be edited directly. To change the EQ settings in this graph, use the two or three knobs below, as explained in the following sections. If, however, you would like to edit the EQ curves graphically, you can do so in the EQ tab (Figure 9-10 on page 66).

### EQ/Dynamics selectors

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

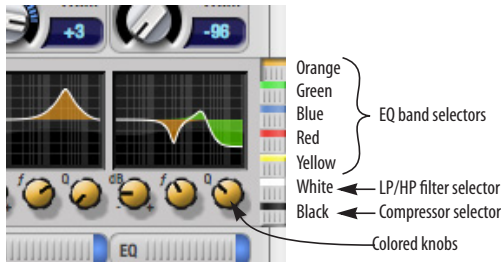


Figure 9-4: The EQ/Dynamics selectors.

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

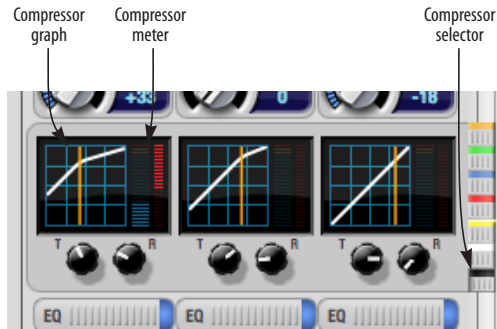


Figure 9-5: The Compressor controls.

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

### Using the EQ/Dynamics knobs

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

☛ **Important:** before you can modify the settings of an EQ band using the three knobs below the graph, the EQ band must be enabled. This is done in the EQ tab (Figure 9-10), as explained in “Enabling EQ” on page 66.

### EQ/Dynamics enable/disable buttons

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

### Reverb send

The input reverb *send* (Figure 9-3) feeds the input signal to the Track16's global reverb processor, where it is merged with any other signals being fed to the reverb. The reverb's output can then be fed back into a mix or output pair. The send occurs after all other settings in the input channel strip (phase invert, EQ, compression, etc.)

## THE OUTPUTS TAB

The Outputs tab (Figure 9-6) lets you apply EQ, dynamics and reverb to any output pair, just before the signal leaves the Track16. This is processing that occurs at the very end of the signal flow, after everything else (host based effects, Track16 input or bus processing, mixing, and so on). Processing is done in the digital domain, just before the signal goes analog through the D/A converter. Output tab processing is applied to the entire output mix (all signals being mixed to the output from various sources).

### Signal flows from top to bottom

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

### Output name

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

## Output channel focus

Click the channel focus button (Figure 9-6) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 9-7 on page 64). Channel focus also determines which channels are being scoped by CueMix’s audio analysis tools, as explained in “Choosing channels for audio analysis” on page 80.

## Output EQ and Dynamics

The EQ/Dynamics section in the Outputs tab (Figure 9-6) works identically to the EQ/Dynamics section for the Inputs tab (Figure 9-3). See “Input EQ and dynamics” on page 61.

## Output reverb send/return

The output reverb *send* (Figure 9-6) feeds the signal for the output to the Track16’s global reverb processor, where it is merged with any other signals being fed to the reverb. The reverb’s output can then be fed back into the mixer at various return points, including the same output from which it was sent (discussed below). The output reverb

send is disabled (grayed out) when the reverb *Split Point* is set to *Mix* to eliminate the possibility for feedback loops created by reverb send/return loops. See “Split point” on page 76.

The output reverb *return* (Figure 9-6) feeds the output of the Track16’s global reverb processor directly to the output. This includes any other signals currently being fed to the reverb.

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

## Talkback/Listenback

Click the *Talkback* or *Listenback* buttons (Figure 9-6) to toggle whether the output pair is included in the Talkback or Listenback group. See “Talkback and listenback” on page 77.

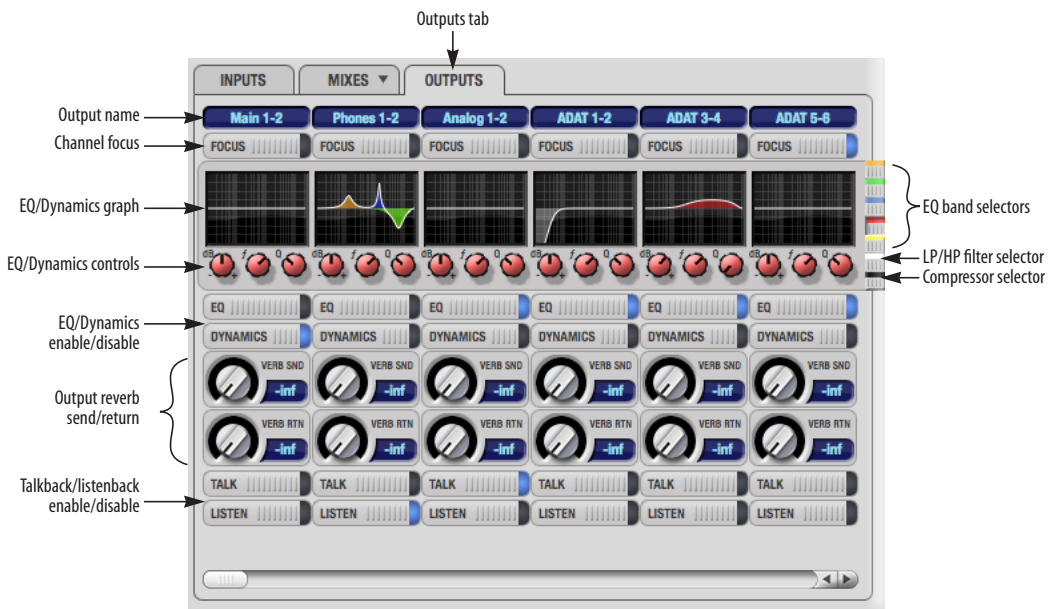


Figure 9-6: The Outputs tab.

## THE CHANNEL SETTINGS SECTION

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

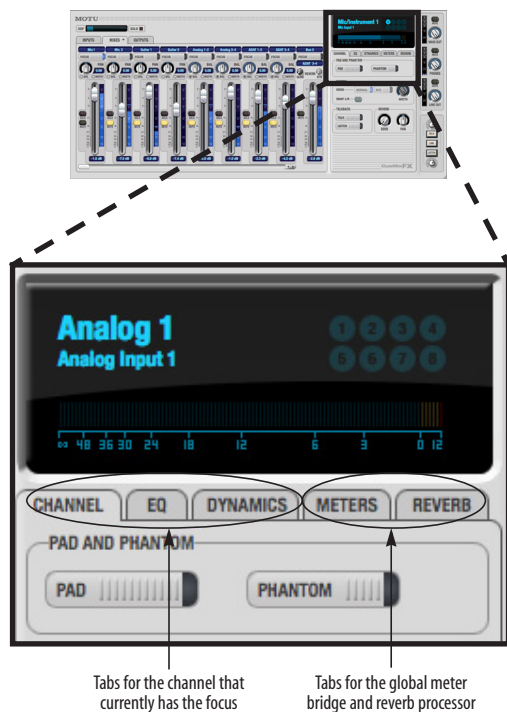


Figure 9-7: The Channel Settings section.

## The Channel tab

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



Figure 9-8: The Channel tab.

## Signal flow

Settings in the Channel tab occur just before the EQ, dynamics and reverb sends in the Input tab channel strip (Figure 9-3 on page 60). Input channel signal flow is as follows: trim, phase, stereo versus M/S decoding, width, L/R swap, EQ, dynamics and reverb send.

## Pad and phantom

The Pad and phantom settings become active (ungrayed) when the focus is on a mic input (or mic input pair). These are the same -20 dB pad and 48V phantom power settings that you can also control from the Track16 hardware itself. There are separate settings for each mic input.



### Stereo settings

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

The *Width* knob (Figure 9-8) provides control over the stereo imaging, going from a full stereo image to mono (both channels panned equally). See “Width” on page 59.

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

### Talkback section

Click the *Talkback* or *Listenback* button (Figure 9-8) to toggle whether the input is the Talkback or Listenback input. Only one input can be the talkback input, and only one input can be the listenback input. See “Talkback and listenback” on page 77.

### Reverb section

The *Send* in the reverb section (Figure 9-8) is the same control as the reverb send in the Input tab channel strip (Figure 9-3). See “Reverb send” on page 62. If the input is currently not grouped as a stereo pair in the Input tab (it is operating as a mono input), use the reverb *Pan* knob (Figure 9-8) to pan the mono signal for the stereo reverb processor.

### Input meter and bus activity LEDs

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

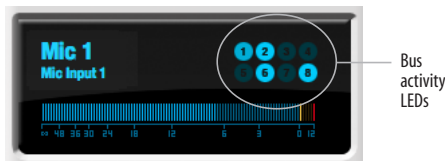


Figure 9-9: Input meter and bus activity LEDs.

The *input level meter* (Figure 9-9) is the same as the input meters in the Meters tab (Figure 9-22 on page 75) with the *Pre FX* button engaged, which shows the input level on the physical input itself, before any processing of any kind occurs within the Track16. This meter gives you the most accurate reading of the actual signal level hitting the input, regardless of any other settings.

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

## The EQ tab

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

## Vintage EQ

Inspired by legendary British large console EQs, the Track16 *Vintage EQ section* (Figure 9-10) gives you the look, feel and sound of the most sought-after classic equalizers. Five bands of center frequency parametric EQ filtering are provided, each with four EQ types that provide current popular EQ styles and vintage analog EQ styles alike. Two bands include shelf filtering. Two additional bands of variable slope low pass and high pass filtering are provided. The filter response display provides comprehensive control and visual

feedback of the EQ curve being applied. With 64-bit floating point processing, the Track16 Vintage EQ has been carefully crafted and meticulously engineered to produce musical results in a wide variety of applications.

## Enabling EQ

Each input and output channel has a global *EQ enable/disable* button (Figure 9-3 and Figure 9-6). This button enables or disables all bands of EQ for the channel. In addition, each individual band of EQ has a *Filter enable/disable* switch (Figure 9-10), allowing you to enable as few or as many bands as needed for each individual channel.

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

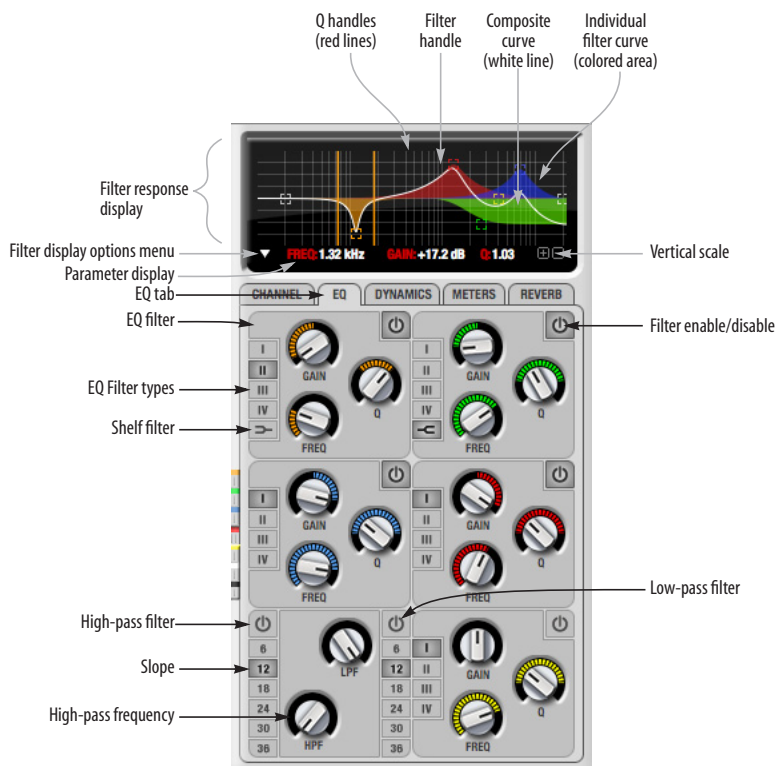


Figure 9-10: The EQ tab.

## Vintage EQ Quick reference

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

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

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

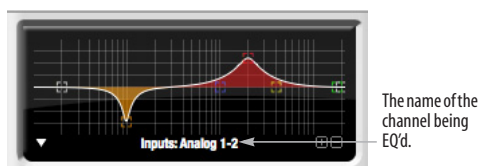


Figure 9-11: When a filter handle is not selected and when the cursor is not hovering over the display, the parameter display shows the name of the current channel being edited (the channel that currently has the focus).

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

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

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

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

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

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

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

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

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

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

## How the vintage EQ works

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

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

## Frequency response display

The frequency response display at the top of the window displays the response curve of the current settings in the window. The (horizontal) frequency range is from 10 hertz to 20 KHz. The (vertical) amplitude scale is in dB and is adjustable between 3 and 24 dB using the *vertical scale* buttons (Figure 9-10).

## Showing and hiding filter curves

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

## Adjusting filters in the display

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

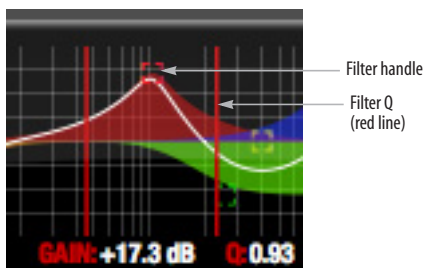


Figure 9-12: Drag the filter handle to adjust its frequency and/or boost/cut. Drag the Filter Q handles to adjust the Q.

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

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

## EQ filters

The EQ filters have three parameters:

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

## Q

The Q setting does not have a unit of measurement. Rather, it is the ratio of the filter's center frequency to the bandwidth of the filter. In addition, the

actual Q value for the EQ curve being applied is dependent on three factors: the gain setting, the filter style, and the Q setting.

## Filter types

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

## Returning to zero (or nominal frequency)

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

## EQ filter styles

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

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

## Type I

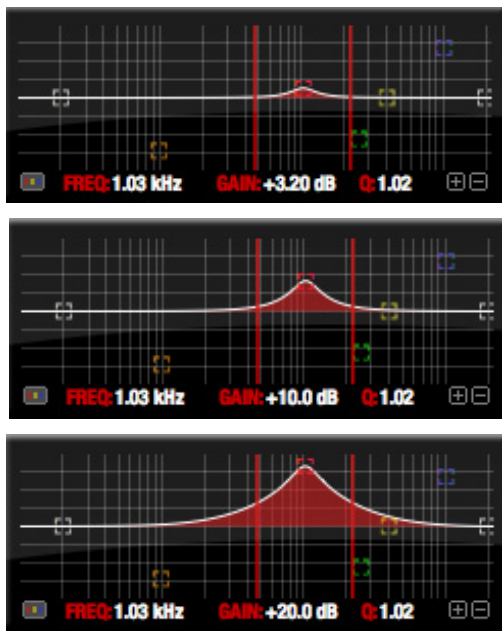


Figure 9-13: Type I EQ filter style.

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

## Type II



Figure 9-14: Type II EQ filter style.

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

### Type III

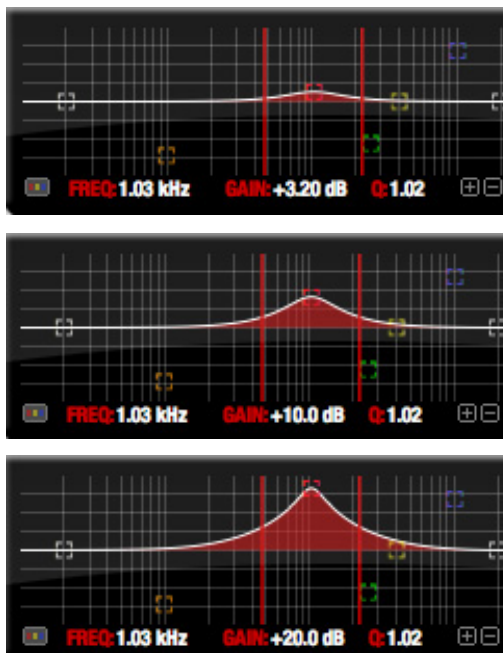


Figure 9-15: Type III EQ filter style.

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

### Type IV

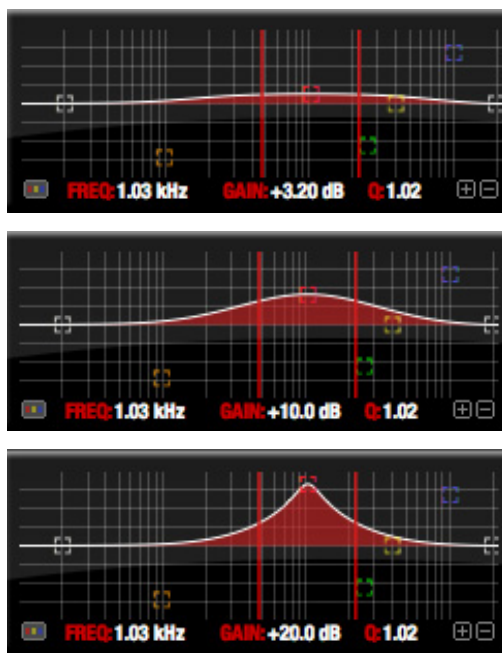


Figure 9-16: Type IV EQ filter style.

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

## Shelf filters

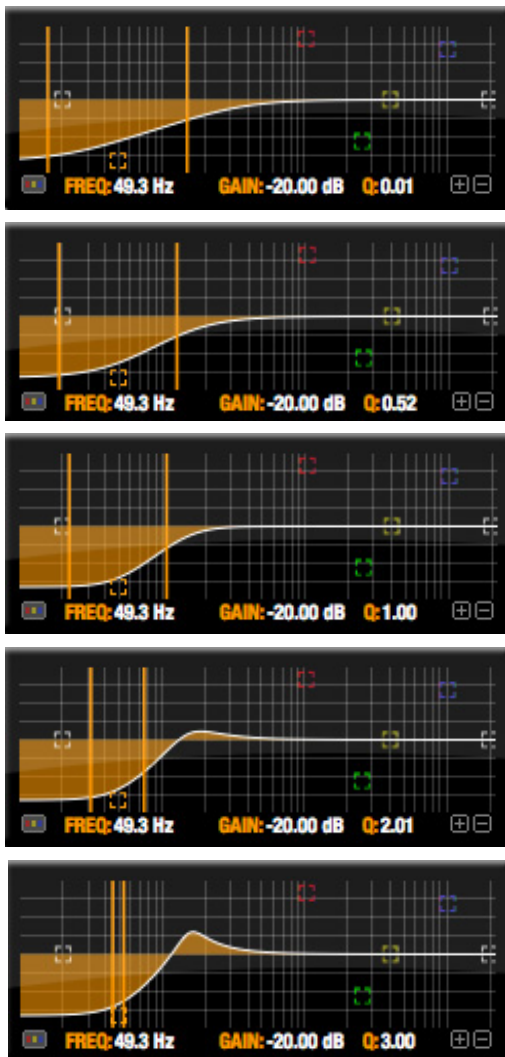


Figure 9-17: Shelf filter Q parameter overshoot.

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

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

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

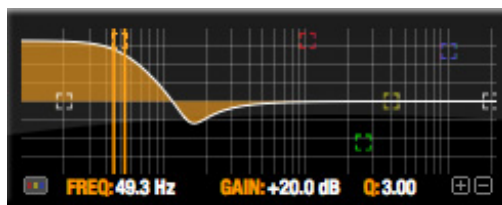


Figure 9-18: Overshoot when low shelf boost is applied.

Overshoot is also applied to high shelf boost and cut:

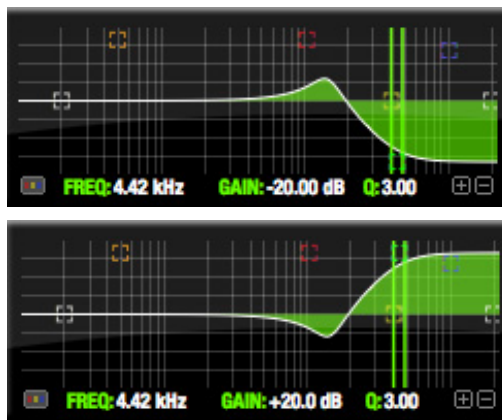


Figure 9-19: Overshoot when high shelf cut and boost is applied.

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

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

### Low pass and high pass filters

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

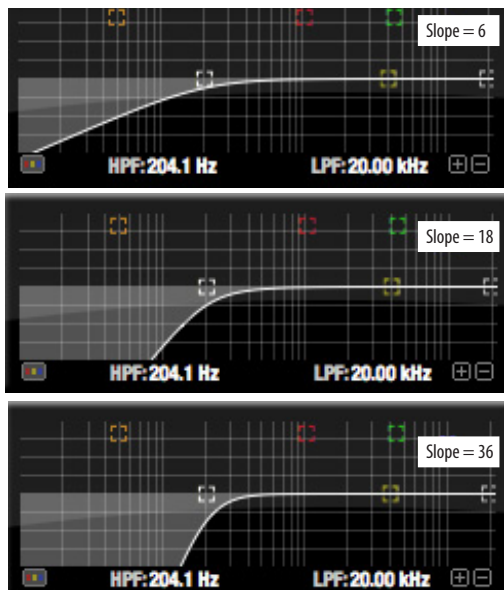


Figure 9-20: The low pass filter with three example slope settings.

### The Dynamics tab

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

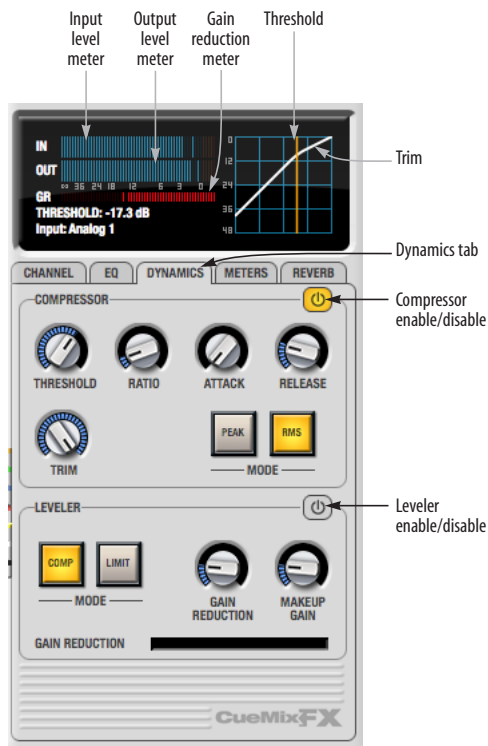


Figure 9-21: The Dynamics tab.

### Enabling Dynamics

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



## Compressor

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

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

### Graphic adjustment of the Threshold

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

### Input level meter

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

### Gain reduction (GR) meter

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

### Output level

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

## Peak/RMS modes

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

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

## Leveler

The Leveler™ (Figure 9-21) provides an accurate model of the legendary Teletronix™ LA-2A® optical compressor, known for its unique and highly sought-after Automatic Gain Control (AGC) characteristics. The Track16 Leveler faithfully models the LA-2A using the on-board DSP with 32-bit floating point precision.

### A model of an optical compressor

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

### Automatic gain control using light

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

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

### Response characteristics

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

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

### Compressor/Limit buttons

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

### Gain Reduction

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

### Makeup Gain

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

### Enabling or disabling the Leveler

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

## The Meters tab

The *Meters* tab (Figure 9-22) serves as a comprehensive meter bridge for all inputs, outputs and mix busses in the Track16. This tab gives you a “bird’s-eye” view of all signal activity in the Track16; it is ideal for confirming your signal routing programming and for troubleshooting.

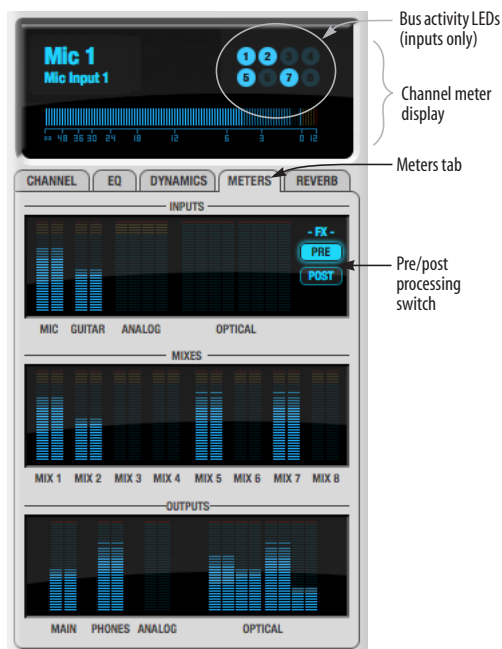


Figure 9-22: The Meters tab.

### Channel meter display

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

### Bus activity LEDs (inputs only)

The *Bus activity LEDs* (Figure 9-22) are present only for inputs. See “Input meter and bus activity LEDs” on page 65.

### Pre/post processing switch

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

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

## The Reverb tab

The *Reverb* tab (Figure 9-23) provides access to the Track16’s single, global reverb processor, which provides high-fidelity reverberation and graphic control over its parameters.

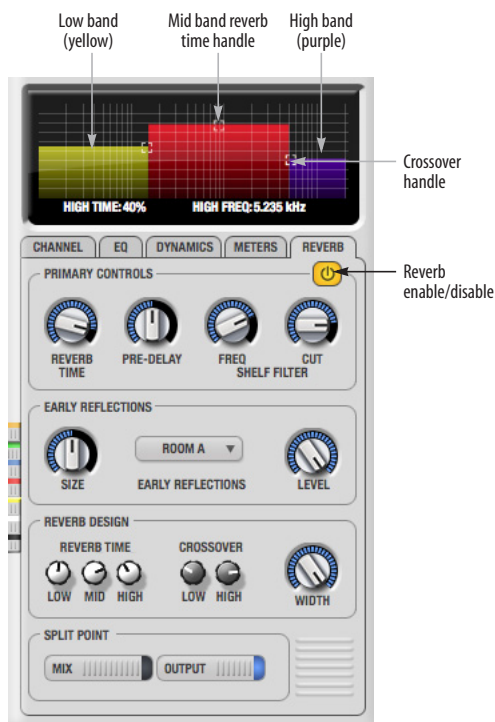


Figure 9-23: The Reverb tab.

### Enabling reverb

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

### Routing inputs, busses and outputs to the reverb processor

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

### Reverb sends

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

- Mono or stereo inputs (Figure 9-3 on page 60)
- Mix bus output (Figure 9-2 on page 58)
- Outputs (Figure 9-6 on page 63)

### Reverb returns

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

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

### Split point

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

### Mix

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

### Output

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

### Primary controls

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

### Reverb Time

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

### PreDelay

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

### Shelf Filter

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

### Early reflections

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

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

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

### Reverb design

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

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

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

### OUTPUT VOLUME CONTROLS

The *Monitoring section* (Figure 9-1) provides independent volume control for each of the Track16's analog outputs.

### DSP METER

The DSP meter (Figure 9-1) shows how much of the available DSP processing power is currently being used by the Track16 for effects processing. DSP resources are allocated in channel order from the first input to the last output. If there aren't enough DSP resources for all effects to be enabled on a channel, none of them are allocated on that channel or any following channel.

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

### SOLO LIGHT

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

### TALKBACK AND LISTENBACK

CueMix FX provides *Talkback* and *Listenback* buttons (Figure 9-1). Talkback allows an engineer in the control room to temporarily dim all audio and talk to musicians in the live room. Conversely, Listenback allows musicians to talk to the control room.

### Hardware setup

Figure 9-24 below shows a typical hardware setup for Talkback and Listenback. For Talkback, set up a dedicated mic in your control room and connect it to a mic input on your MOTU audio interface. For Listenback, set up a dedicated listenback mic in the live room for the musicians and connect it to another mic input (or just use one of the mics you are recording from). For talkback output, set up a headphone distribution amp or set of speakers in the live room, and connect it to any Track16 output, as demonstrated below in Figure 9-24.

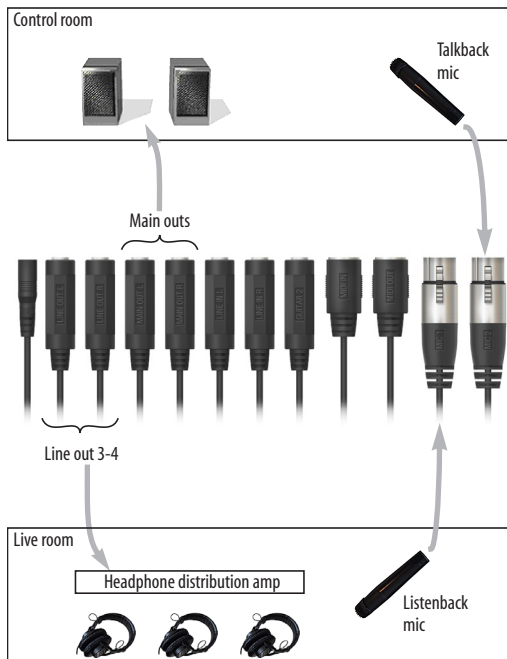


Figure 9-24: Typical hardware setup for Talkback and Listenback.

### Talkback / Listenback Mic Input

To configure the talkback mic in CueMix FX, go to the Inputs tab (Figure 9-3 on page 60) and click the Focus button for the input that the talkback mic is connected to. Click the Channel tab (Figure 9-8 on page 64) and enable the *Talk* button.

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

### Talk / Listen output

To configure the talkback and listen back outputs, go to the Outputs tab (Figure 9-6 on page 63) and enable the *Talk* button for any output pair on which you'd like to hear the talkback mic. Similarly, enable the *Listen* button for any output pair on which you'd like to hear the listenback mic.

### Talkback / Listenback Monitor Dim

Use the knobs next to the Talk and Listen buttons (Figure 9-25) to determine the amount of attenuation you would like to apply to all other

audio signals (besides the talkback/listenback signal) when Talkback and/or Listenback is engaged. To completely silence all other CueMix audio, turn them all the way down. Attenuation only occurs when talkback or listenback is engaged. Audio playing back from disk (your host software) is not affected.



Figure 9-25: The Talkback/Listenback controls.

### Engaging/disengaging Talkback and Listenback

To engage Talk back or Listenback, press on the Talk or Listen buttons (Figure 9-25) and then release to disengage. Talkback and/or Listenback is engaged for as long as you hold down the mouse button. Alt-click to make the buttons “sticky” (stay engaged until you click them again — so you don't have to hold down the mouse). Or use the Talkback menu items.

If you would like to engage both Talkback and Listenback at the same time, enable the *Link* button (Figure 9-25).

### Controlling Talkback and Listenback volume

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

## SHORTCUTS

Hold down the following general modifier keys as shortcuts:

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

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

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

## FILE MENU

### Saving and loading hardware presets

The Track16 can store up to 16 presets in its on-board memory. A preset includes all CueMix FX settings for all mix busses, but it excludes global settings like clock source and sample rate.

The Load Hardware Preset and Save Hardware Preset commands in the CueMix FX file menu let you name, save and load presets in the Track16.

### Peak/hold time

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

### Mix1 Return Includes Computer

The *Mix1 return includes computer* File menu item applies to other MOTU interfaces products and has no effect on the Track16.

### Hardware follows CueMix Stereo Settings

This File menu item applies to other MOTU interfaces products and has no effect on the Track16.

## EDIT MENU

### Undo/Redo

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

### Copying & pasting (duplicating) entire mixes

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

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

### Clear Peaks

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

## DEVICES MENU

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

### Audio analysis tools

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

FFT and Spectrogram display .....	81
Oscilloscope.....	83
X-Y Plot.....	89
Phase Analysis .....	92
Tuner.....	96

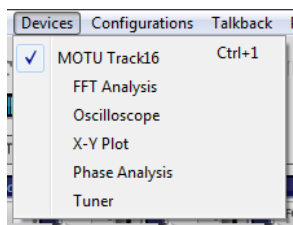


Figure 9-26: Opening the signal analysis windows.

### Choosing channels for audio analysis

The audio analysis tools follow the currently focused audio input or output. (See “Channel focus and settings” on page 57.) If you focus a mono channel (e.g. Analog 3), its corresponding stereo pair will be displayed (Analog 3–4).

### Scoping host software audio output

If you want to scope audio output from your host software, send your host’s output to an Track16 output pair, and then set the focus on that output pair.



## FFT AND SPECTROGRAM DISPLAY

FFT and spectrogram information can be displayed in the Filter response display section in the EQ tab (Figure 9-10 on page 66) or as a separate window (Figure 9-28 on page 81) opened from the Devices menu (Figure 9-26).

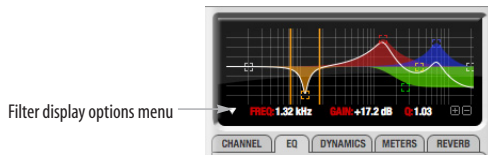


Figure 9-27: Filter Display options menu

### Filter display options

The *Filter display options* menu (Figure 9-27) provides several options for the EQ filter display:

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

## FFT display

Choose *Show FFT* from the *Filter display options* menu (Figure 9-10) to superimpose a real-time Fast Fourier Transform (FFT) frequency measurement curve over the EQ filter display, as demonstrated in Figure 9-29:

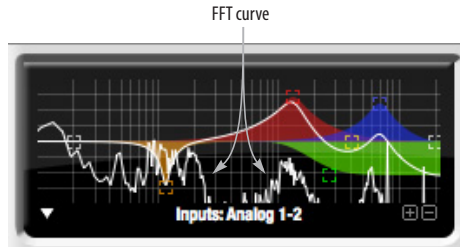


Figure 9-29: FFT display.

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

## Spectrogram

Choose *Show Spectrogram* from the *Filter display options* menu (Figure 9-10) to superimpose a real-time spectrogram “waterfall” display in the background of the EQ filter display, as demonstrated in Figure 9-30:

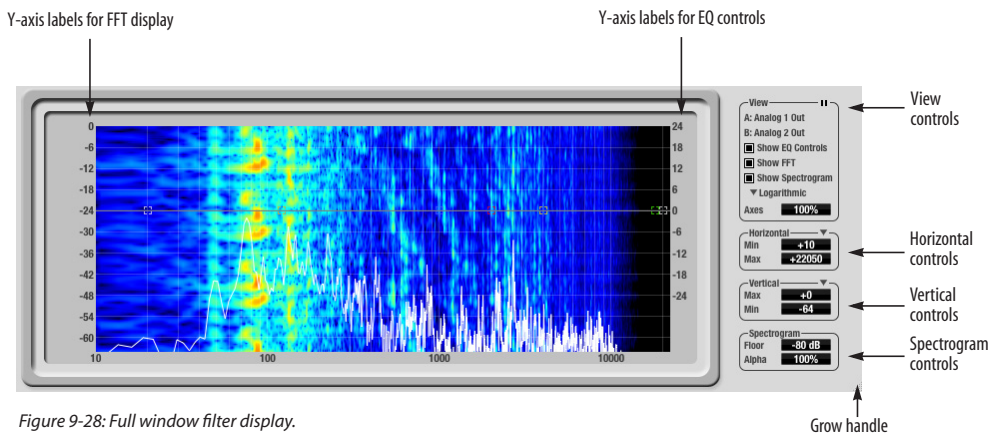


Figure 9-28: Full window filter display.

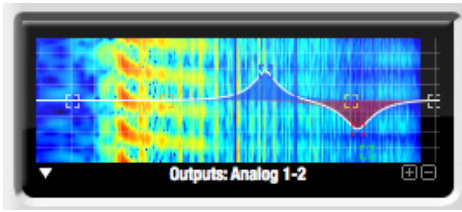


Figure 9-30: FFT display.

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



Figure 9-31: Spectrogram color-to-amplitude spectrum.

### Opening the FFT Analysis window

Choose *FFT Analysis* from the *Devices* menu (Figure 9-26) to open a new window with the filter EQ display for detailed inspection and adjustment of the EQ filter, as shown Figure 9-28.

### View controls

You can show and hide the FFT display, spectrogram or EQ band response curves as desired using the *View* controls (Figure 9-32).

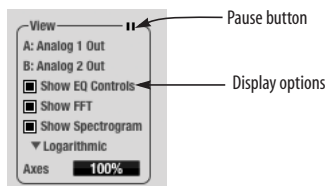


Figure 9-32: View controls

These settings are independent of the small graph display options (Figure 9-10 on page 66), so you have the flexibility to display different combinations in each graph.

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

### Logarithmic or Linear X-Axis Scale

The x-axis defaults to a logarithmic scale, but can be changed to a linear scale if desired. In the *View* controls (Figure 9-32), click *Logarithmic* to access the x-axis scale options menu. With a linear scale selected, frequency is constant, but the width of each octave along the x-axis is different. With a logarithmic scale selected, octaves are displayed with a constant width, but frequency is displayed logarithmically within each octave.

### Axes display

The *Axes* control (Figure 9-32) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

### Pausing the display

The *Pause* button in the upper right corner of the *View* section (Figure 9-32) allows you to freeze the display at any time. To resume, click the button again.

### Horizontal controls (frequency axis)

The *Horizontal* controls (Figure 9-33) configure the value range of the x-axis (frequency). Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the *Horizontal* control menu (Figure 9-33).

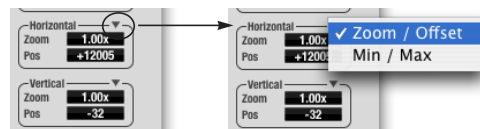


Figure 9-33: Horizontal control menu

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

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

### Vertical controls (amplitude axis)

The *Vertical* controls (Figure 9-28) operate similarly to the Horizontal controls, except that they configure the y-axis (amplitude).

In *Zoom/Offset* mode, *Zoom* sets the display zoom from 1x to 100x, and *Pos* sets the center amplitude of the graph. In *Min/Max* mode, *Min* and *Max* set the smallest and largest displayed amplitude.

### Spectrogram controls

The *Floor* control (Figure 9-28) sets the amplitude threshold for the spectrogram display, from -144 dB up to 0 dB.

The *Alpha* control (Figure 9-28) sets the opacity of the spectrogram information displayed in the graph, from 100% (fully visible) to 0% (hidden).

### The info box

When any EQ filter point is selected or dragged in the full window graph, the info box is shown next to the point in the full graph display (Figure 9-35).

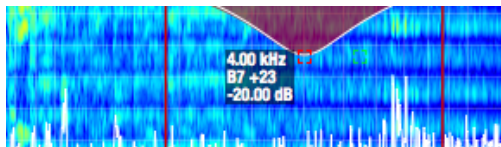


Figure 9-35: The Info Box.

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

### OSCILLOSCOPE

The Oscilloscope (Figure 9-34) graphs the amplitude of an audio signal over time.

Amplitude is displayed on the y-axis and time is displayed on the x-axis. A thick white vertical line marks where time equals zero; a thick white horizontal line marks where amplitude equals zero (Figure 9-34, below).

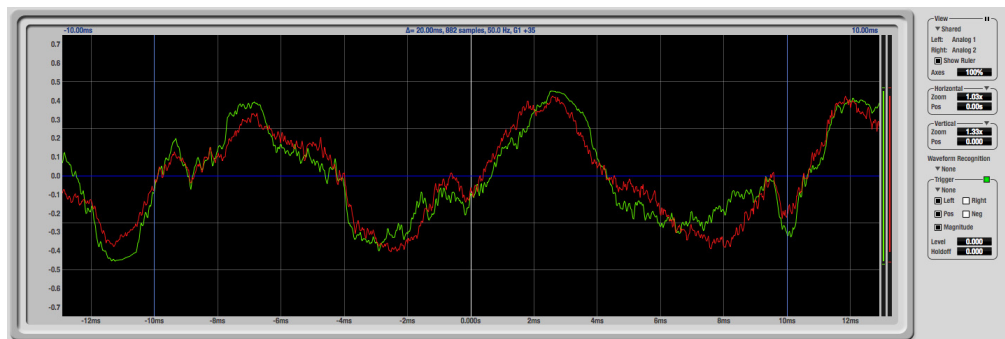


Figure 9-34: Oscilloscope

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

### Opening the oscilloscope

Each Track16 has its own oscilloscope. To open an oscilloscope, choose the *Oscilloscope* item from the Devices menu under the desired interface.

### Choosing a channel to display

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

### View controls

The View controls (Figure 9-36) provide several options for the oscilloscope display.

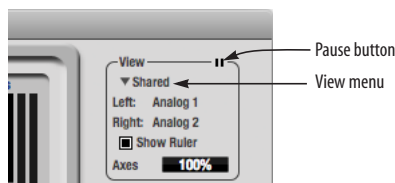


Figure 9-36: View controls

### View menu

The View menu (Figure 9-36) lets you choose how you wish to display the audio channel(s) being displayed.

#### View menu setting What it displays

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

### Display options

The *Axes* control (Figure 9-36) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden). The *Show Ruler* option toggles the measurement items (see “Measurement information” on page 87).

### Pausing the display

The Pause button in the upper right corner of the View section (Figure 9-36) allows you to freeze the display at any time. To resume, click the button again. The level meters will remain active while the display is paused.

### Horizontal controls (time axis)

The *Horizontal* controls (Figure 9-37) configure the value range of the x-axis (time). Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the Horizontal control menu (Figure 9-37).

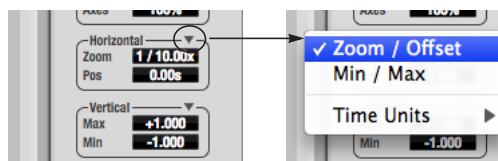


Figure 9-37: Horizontal control menu

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

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

## Time Units

The *Time Units* sub-menu (Figure 9-37) provides the option to view the X axis in Seconds or Samples.

## Vertical controls (amplitude axis)

The *Vertical* controls (Figure 9-37) operate similarly to the Horizontal controls, except that they configure the y-axis (amplitude).

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

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

## Waveform Recognition

The Waveform Recognition option searches through new audio data looking for a waveform which most resembles that which was previously displayed. The region where this takes place is a small window around the line marking time equals zero, denoted by the extra vertical graph lines surrounding it. There are two kinds of waveform recognition available: Type I and Type II.

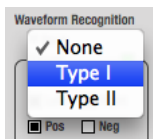


Figure 9-38: Waveform Recognition menu

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

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

## Trigger

When the *Trigger* (Figure 9-39) is not enabled (the Trigger menu is set to *None*), the graph updates based on time: after every *n* samples of the monitored audio signal, the most recent samples are displayed. When the Trigger is enabled (set to any mode other than *None*), the graph updates in response to specific conditions in the signal. The Trigger section defines that criteria and how the graph will display the events that match.

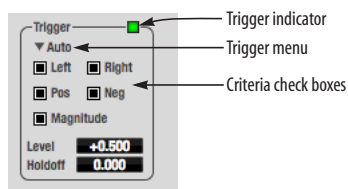


Figure 9-39: Trigger settings

## Criteria

The criteria checkboxes (Figure 9-39) determine the conditions that the trigger is looking for and where it will look for them.

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

The *Pos* and *Neg* checkboxes determine the slope of the event. When the *Pos* checkbox is enabled, the trigger will look for an event where amplitude is increasing; likewise, enabling the *Neg* checkbox tells the trigger to look for an event where amplitude is decreasing. One or both of these can

be enabled simultaneously. If neither is enabled, the criteria will not be found because the trigger is not looking for any particular kind of event.

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

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

### Holdoff

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

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

### Trigger modes

The Trigger menu (Figure 9-39 on page 85) provides four modes:

#### Trigger mode What it does

None	The Trigger is not active; this is the default mode. The incoming audio signal will be displayed continuously as audio is received.
Auto	The display is always updating, but when the condition is met, the trigger event will be displayed centered around the line marking time equals zero.
Normal	The display updates only when the condition is met; the last trace will be displayed until the next matching event is found.
Single Sweep	Similar to Normal mode, but the last trace will be displayed until you manually arm the trigger by clicking the Trigger indicator (Figure 9-39 on page 85) or by pressing the spacebar.

### Trigger indicator

The Trigger indicator (Figure 9-39 on page 85) displays the state of the trigger, and also provides a way to manually interact with it. The Trigger indicator always displays one of three colors:

Color	Status
Green	When the current Trigger criteria has been met (including when the Trigger mode is <i>None</i> ).
Yellow	When the Trigger is armed, but has not yet found an event which matches its criteria. Yellow can also indicate that the graph has been manually paused using the Pause button in the View section (see “Pausing the display” on page 84).
Red	When the Trigger is being held off, either because the Trigger mode is set to Single Sweep or the Holdoff time is not set to zero.

You can also click on the Trigger indicator to force certain actions, depending on the Trigger mode. In Auto and Normal modes, clicking on the Trigger indicator causes the display to run freely; you may click & hold to force this to occur for as long as you'd like. In Single Sweep mode, clicking on the

Trigger indicator re-arms the trigger. When the Trigger mode is *None*, clicking on the Trigger indicator has no effect.

### Measurement information

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

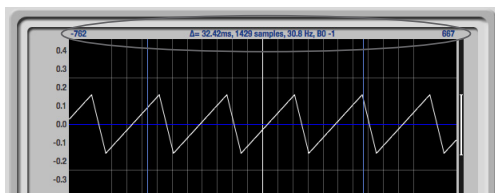


Figure 9-40: Measurement information

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

Information about the measured area is displayed at the center of the top ruler: the duration (in seconds and samples), the approximate frequency,

and the scientific note name. If the measured area is long enough, the approximate beats per minute (bpm) is displayed.

### Ideas for using the Oscilloscope

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

#### Analyzing and comparing harmonic content

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

#### Viewing transients such as drum hits

If you loop a snare hit or other similar transient audio clip and feed it through the oscilloscope, you can more or less “freeze” the transient waveform in the oscilloscope frame. This can be useful, for example, for viewing the results of real-time compression that you are applying with an effects plug-in, as demonstrated in Figure 9-41. In this example, a snare hit is being compressed by Digital Performer’s Dynamics plug-in. As you make

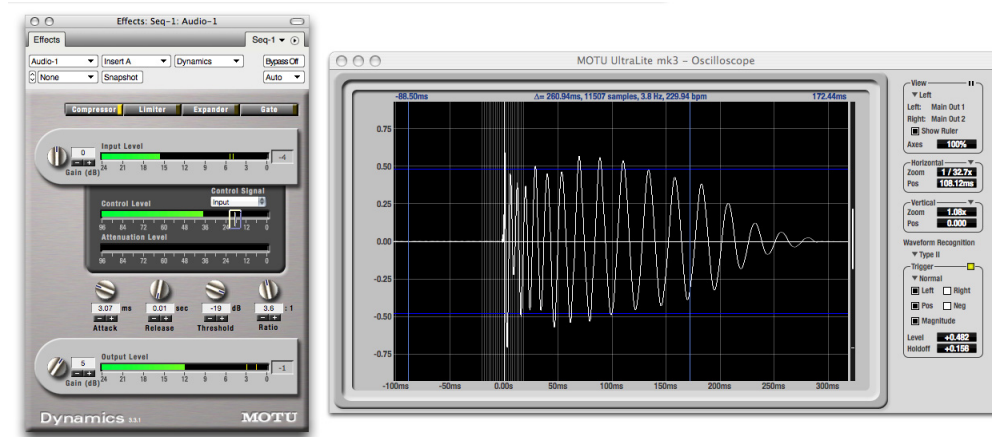


Figure 9-41: Viewing transients in the Oscilloscope

adjustments to the compression plug-in's settings, you can see the transient waveform change the next time the Oscilloscope triggers. For compression, this can be particularly useful for balancing the effect of the attack on the transient, relative to the decay portion of the waveform. Conversely, you can see the effect of the threshold setting directly on the decay portion, relative to the attack. In effect, you can see as well as hear the results of your compression adjustments.

To view a transient waveform in the Oscilloscope display, turn off Waveform Recognition and use the *Normal Trigger* mode. Adjust the level high enough to encompass the vertical amplitude of most of the transient. If the transient pulse sweeps across the screen, try raising the Holdoff level. Once the transient is settled in the display and fairly stable, you may need to adjust the horizontal position to center it in the display. These settings are depicted in the example in Figure 9-41.

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

### Clip detection

You can use the Oscilloscope to detect clipping in a digital audio signal. To do so, enable all criteria (Figure 9-39 on page 85), choose *Single Sweep* from the trigger menu (Figure 9-39), set the level to 0.999 and click the trigger indicator (Figure 9-39) to arm it (yellow). As soon as the signal clips, the trigger indicator will turn red, and the display will show the offending clip at the line marking time equals zero.

### Viewing timing pulses

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

### Building synthesizer patches

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

Another example is waveform polarity. If you are combining several raw waveforms, polarity is a critical, yet not always obvious, factor in determining the resulting sound. You can use the Oscilloscope to easily view and compare polarities to see if they are inverted from one another or not. The *Add* and *Subtract L - R* View menu settings are particularly useful here.

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

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

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



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

For more information on Volta, see [www.motu.com](http://www.motu.com).

### X-Y PLOT

The *X-Y Plot* window (Figure 9-42) graphs the amplitude of a stereo audio signal on a two-dimensional grid.

For each unit of time (i.e., each sample), the amplitude of the left channel is displayed on the x-axis and the amplitude of the right channel is displayed on the y-axis. A thick white vertical line marks where left channel amplitude equals zero; a thick white horizontal line marks where right channel amplitude equals zero (Figure 9-42, below). There are also thick white diagonal lines for  $y = x$  and  $y = -x$ .

### Metering

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

The higher the meter, the higher the correlation between the two channels. Below are a few examples:

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

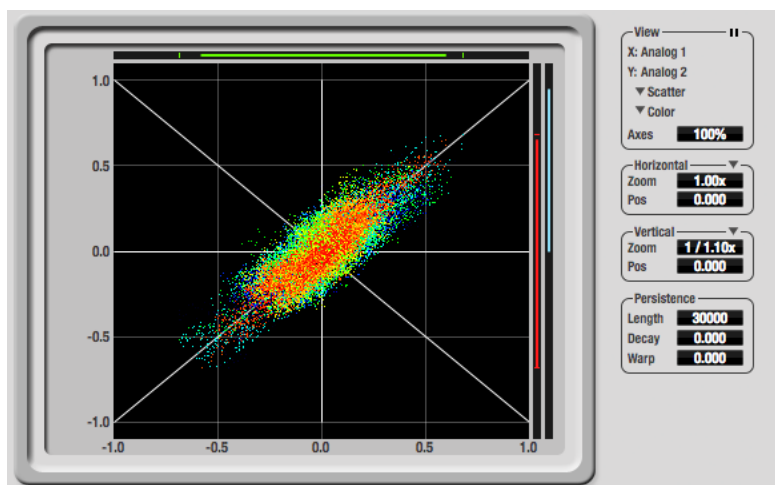


Figure 9-42: X-Y Plot

## Opening the X-Y Plot

Each Track16 interface has its own X-Y Plot window. Choose the *X-Y Plot* item from the Devices menu under the desired interface.

## Choosing a channel pair to display

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

## View controls

The View controls (Figure 9-43) provide several options for the X-Y Plot display.

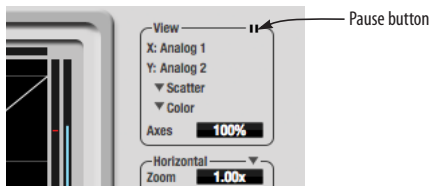


Figure 9-43: View controls

## Pausing the display

The Pause button in the upper right corner of the View section (Figure 9-43) allows you to freeze the display at any time. To resume, click the button again. The level meters will remain active while the display is paused.

## Line/Scatter

Choose either *Line* or *Scatter* from the menu in the View section (Figure 9-43) to plot each point (sample) as either a single pixel or as a continuous line that connects each plot point to the next, as shown below in Figure 9-44.

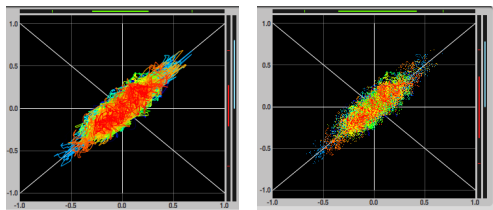


Figure 9-44: The same X-Y Plot displayed in Line versus Scatter mode.

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

## Color/Grayscale

In *Color* mode (Figure 9-43) the most recently displayed audio data is shown in red, which fades to yellow, green and then finally blue, before disappearing. In Grayscale mode, data is first shown in white and then fades to gray. To adjust the scale of this color/brightness change, see “Decay” on page 91.

## Axes

The *Axes* control (Figure 9-43) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

## Horizontal and vertical controls

The *Horizontal* and *Vertical* controls (Figure 9-45) configure the value range of the x-axis (left channel amplitude), and y-axis (right channel amplitude), respectively. Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the menu shown in Figure 9-45.

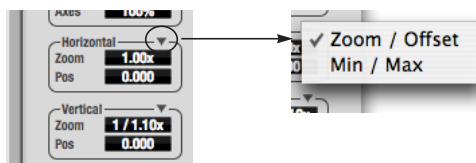


Figure 9-45: Setting the Horizontal or Vertical control modes.

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

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

## Persistence

The *Persistence* controls (Figure 9-46) affect the appearance of data from when it is first displayed until it disappears from the grid.

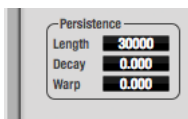


Figure 9-46: The Persistence controls.

## Length

*Length* (Figure 9-46) sets the number of recent samples to show on the plot. For example, when *Length* is set to 10,000, the 10,000 most recent samples are shown.

## Decay

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

*Decay* (Figure 9-46 on page 91) determines the brightness or hue of the minimum value. When *Decay* is zero, the oldest sample is black. When *Decay* is +1.000, the oldest sample is fully opaque (in Grayscale mode) or red (in Color mode).

## Warp

*Warp* (Figure 9-46) determines the position of data points after they are first drawn. When *warp* is zero, data points remain in the same position.

When *warp* is positive, they contract towards the origin (center of the grid). When *warp* is negative, they expand away from the origin. The further the *warp* value is from zero, the greater the effect.

## Using the X-Y Plot

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

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

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

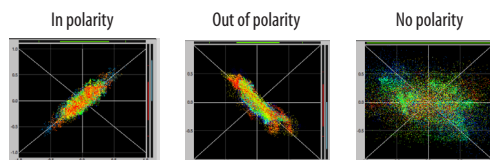


Figure 9-47: Checking polarity in a stereo signal with the X-Y Plot.

## PHASE ANALYSIS

The *Phase Analysis* window (Figure 9-48 on page 92) graphs frequency versus phase difference versus amplitude of a stereo signal on either rectangular or polar coordinates.

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

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

### Correlation Meter

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

### Opening the Phase Analysis

Each MOTU audio interface has its own Phase Analysis window. Choose the *Phase Analysis* item from the Devices menu under the desired interface (Figure 9-26 on page 80).

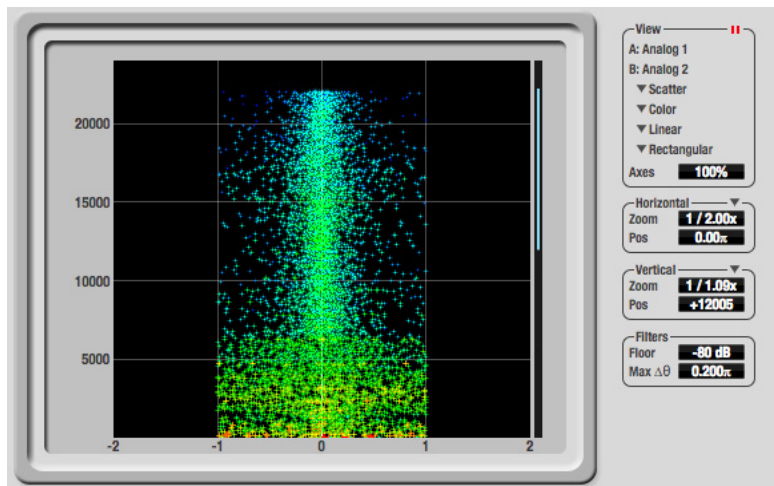


Figure 9-48: Phase Analysis

### Choosing a channel pair to display

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

### View controls

The View controls (Figure 9-49) provide several options for the Phase Analysis display.

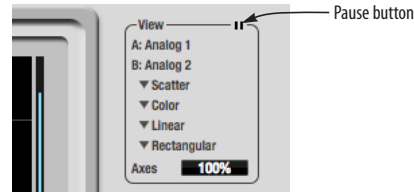


Figure 9-49: View controls

### Pausing the display

The Pause button in the upper right corner of the View section (Figure 9-49) allows you to freeze the display at any time. To resume, click the button again. The correlation meter will remain active while the display is paused.

### A/B (stereo audio channels)

The *View* section (Figure 9-49) displays the pair of input or output audio channels you are viewing. See “Choosing a channel pair to display” above.

### Line/Scatter

Choose either *Line* or *Scatter* from the menu in the *View* section (Figure 9-49) to plot each data point as either a single pixel or as a continuous line that connects each frequency data point to the next, as shown below in Figure 9-44.

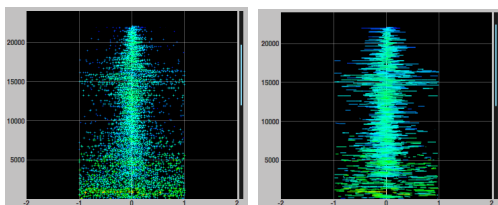


Figure 9-50: The same Phase Analysis displayed in Line versus Scatter mode.

☛ Line mode is significantly more CPU intensive than Scatter. You can reduce Line mode CPU overhead for the Phase Analysis display by increasing the Floor filter and reducing the Max Delta Theta filters (see “Filters” on page 94).

### Color/Grayscale

In *Color* mode (Figure 9-49) signal amplitude is indicated by color as follows: red is loud and blue is soft. In grayscale mode, white is loud and gray is soft.

### Linear/Logarithmic

Choose either *Linear* or *Logarithmic* from the menu in the *View* section (Figure 9-49) to change the scale of the frequency axis. In rectangular coordinates, the vertical axis represents frequency, and in polar coordinates, the radius from the center is frequency. With a linear scale, frequencies are spaced evenly; in a logarithmic scale, each octave is spaced evenly (frequencies are scaled logarithmically within each octave).

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

### Rectangular/Polar

Choose either *Rectangular* or *Polar* from the menu in the *View* section (Figure 9-49) to control how audio is plotted on the Phase Analysis grid.

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

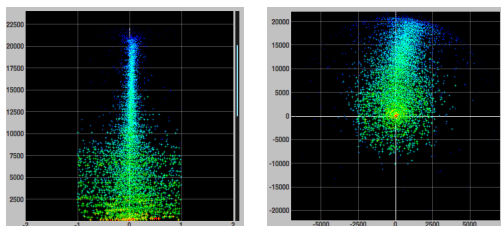


Figure 9-51: Rectangular versus Polar display (with a linear plot).

Above, Figure 9-51 shows Rectangular versus Polar display with a Linear plot. Below, Figure 9-52 shows the same displays (and the same data) with a Logarithmic plot:

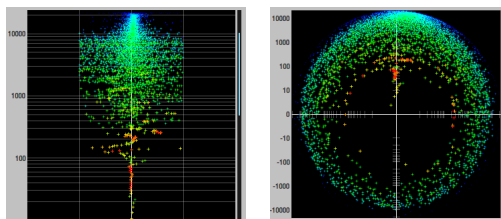


Figure 9-52: Rectangular versus Polar display with a logarithmic plot.

### Axes

The *Axes* control (Figure 9-49) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

## Horizontal and vertical controls

The *Horizontal* and *Vertical* controls (Figure 9-53) let you scale each axis of the grid and offset its zero point. Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the menu shown in Figure 9-53.

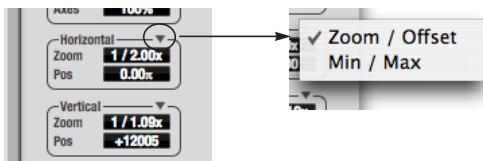


Figure 9-53: Setting the Horizontal or Vertical control modes.

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

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

## Filters

The *Filters* section (Figure 9-54) lets you control the density of the Phase Analysis display.

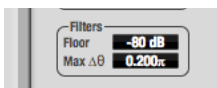


Figure 9-54: Filters

## Floor

*Floor* (Figure 9-54) determines the amplitude threshold for the display. When the amplitude of both channels drops below this threshold, the signal is not shown.

## Max delta theta

*Max delta theta* (Figure 9-54) only affects Line view (see “Line/Scatter” on page 93) and sets the maximum difference in frequency between plot points in the line plot. For two adjacent

frequencies, if the distance (phase difference) between the two frequencies is greater than the Max delta theta, then the line is not drawn.

## Using the Phase Analysis

In the polar display (top row of Figure 9-55 on page 95), stereo material that is predominantly phase-aligned (correlated) appears along the vertical axis, as demonstrated in the first column (*Perfectly in phase*) in Figure 9-55. If the vertical line tilts left or right, this indicates general differences in phase; the more the tilt (delta theta), the more the phase difference. If the vertical line points downwards in the polar display, this indicates that the stereo image is predominantly out of polarity, as demonstrated by the fourth column (*Inverted*) in Figure 9-55. Delays appear as spirals in the polar display.

The rectangular display (bottom row of Figure 9-55) also shows a predominantly phase-aligned stereo image along the vertical axis, and tilt (or left-right offset) from the center vertical axis represents differences in phase. If a signal is predominantly out of polarity, it appears along the  $\theta = -1.0$  or  $\theta = +1.0$  lines in the rectangular display, as demonstrated in the fourth column (*Inverted*) in Figure 9-55 on page 95.

## Using Phase Analysis for multiple mic placement

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

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

### Tuning PA systems

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

### Summing to mono

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

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

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

### Checking for phase issues in stereo tracks

You can use the Phase Analysis window to check the overall polarity of a stereo mix. Figure 9-56 is an example of a full stereo mix that has phase issues, as indicated by the majority of the signal's energy, which is predominantly skewed to the left side of the rectangular view (left) and spread along the -y axis in the polar view (right).

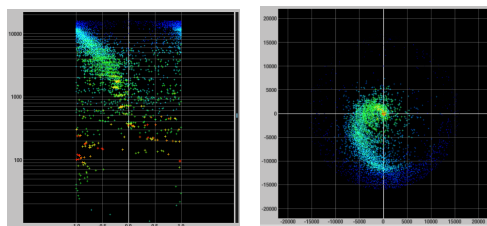


Figure 9-56: A stereo mix with phase issues.

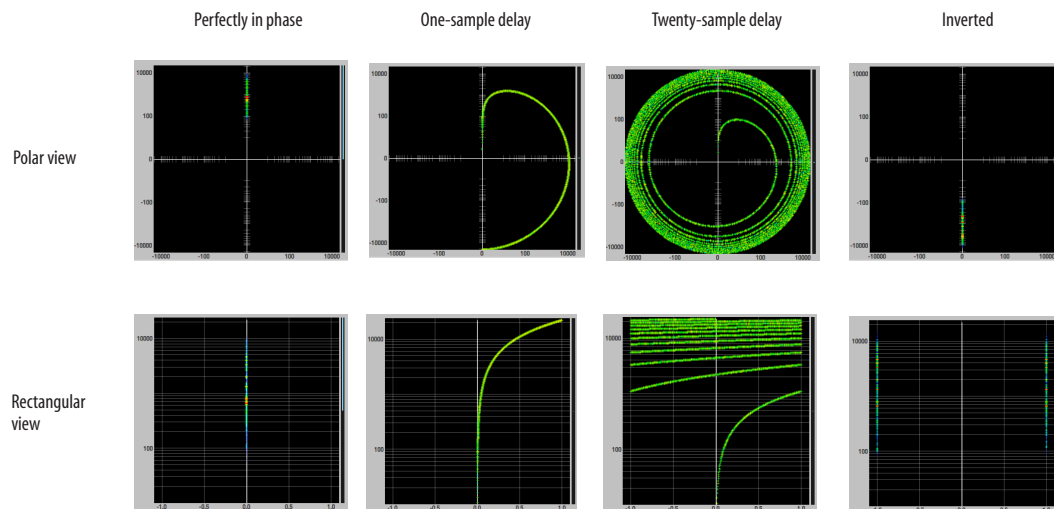
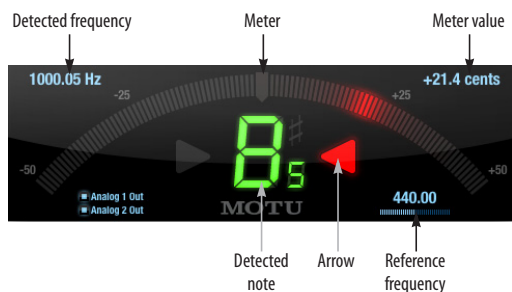


Figure 9-55: Two identical audio streams in the Phase Analysis.

## TUNER

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



### Opening the Tuner

Each MOTU audio interface has its own Tuner window. Choose the *Tuner* item from the Devices menu under the desired interface (Figure 9-26 on page 80).

### Choosing a channel to tune

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

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

### Tuner controls

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

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

**Meter:** representation of the pitch difference between the detected note and the detected fundamental frequency. The horizontal position of the illuminated segments indicates how far the detected frequency is from the detected note. The number of illuminated segments indicates uncertainty or inharmonicity in the signal; a

greater number of illuminated segments represents greater uncertainty. The color of the segments changes gradually from green (in tune) to yellow, orange, and red (progressively further out of tune).

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

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

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

### Tuning stereo signals

When tuning a stereo signal, the tuner analyzes the sum of the two channels. If the channels are not phase coherent, the tuner may not be able to measure the frequency of the signal. To tune only one channel of the channel pair, disable one of the channels as described in “Choosing a channel to tune” on page 96.



## CONFIGURATIONS MENU

A configuration is just like a hardware preset (a “snapshot” of all settings in CueMix FX and therefore the Track16 hardware itself), except that it can be created and managed using the CueMix FX software on your computer, completely independently of the Track16 hardware. The commands in the Configurations menu let you create, save, load, import, export and otherwise manage as many configurations as you wish.

Here is a summary of Configurations menu operations:

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

### Modifying a configuration

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

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

### Saving a CueMix FX configuration as a hardware preset

To save a CueMix FX configuration as a hardware preset:

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

### Saving a hardware preset as a CueMix FX configuration

To save a hardware preset as a CueMix FX configuration:

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

## TALKBACK MENU

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

## PHONES MENU

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

## CONTROL SURFACES MENU

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

### Application follows control surface

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

### Share surfaces with other applications

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

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

## CueMix Control Surfaces

CueMix FX includes support for the following control surface products:

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

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

### Enabled

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

### Configure...

Choose this menu item to configure your control surface product. Open the help files for specific, detailed instructions on configuring CueMix FX for operation with your control surface product.

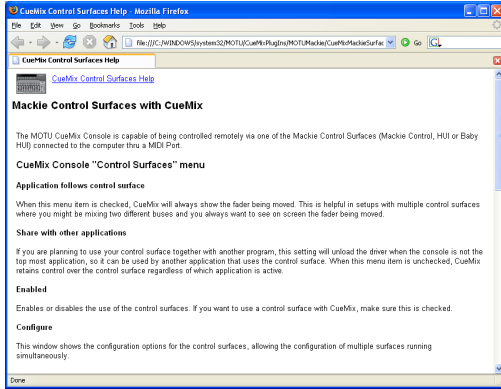
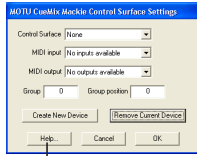


Figure 9-57: Refer to the extensive on-line help for details about configuring CueMix FX for operation with your control surface product.



# CHAPTER 10 MOTU SMPTE Console

## OVERVIEW

The Track16 can resolve directly to SMPTE time code via any line input, without a separate synchronizer. The Track16 can also generate time code via its time code output. The Track16 provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy. Direct time code synchronization is supported by Cubase, Nuendo and other audio sequencer software that supports the ASIO2 sample-accurate sync protocol.

MOTU SMPTE Console .....	101
Clock/Address .....	101
Frame Rate .....	101
Reader section .....	102
Generator section .....	103
Resolving Cubase or Nuendo to time code .....	103
Setup for SMPTE time code sync .....	104

## MOTU SMPTE CONSOLE

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

## CLOCK/ADDRESS

The *Clock/Address* menu (Figure 10-1) provides the same global *Master Clock Source* setting as in MOTU Audio Console (“Master Clock Source” on page 34), but it includes additional information: each setting shows both the clock and the address (time code or sample location), separated by a forward slash (/). To resolve the Track16 to SMPTE time code, choose the *SMPTE / SMPTE* setting in the *Clock/Address* menu. This means that the system will use SMPTE as the clock (time base) and SMPTE as the address.

## FRAME RATE

This setting should be made to match the SMPTE time code frame rate of the time code that the system will be receiving. The Track16 can auto-

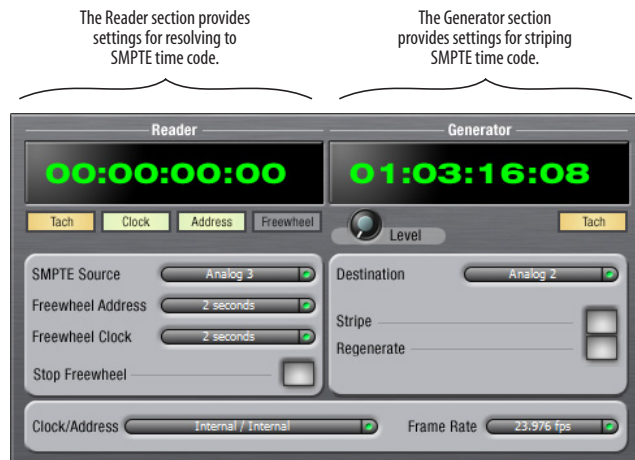


Figure 10-1: SMPTE Console gives you access to your Track16's on-board SMPTE time code synchronization features.

detect and switch to the incoming frame rate, except that it cannot distinguish between 30 fps and 29.97 fps time code, or 23.976 and 24 fps time code. So if you are working with either of these rates, make sure you choose the correct rate from this menu.

## READER SECTION

The Reader section (on the left-hand side of the window in Figure 10-1) provides settings for synchronizing the Track16 to SMPTE time code.

### Status lights

The four status lights (Tach, Clock, Address and Freewheel) give you feedback as follows.

#### Tach

The Tach light blinks once per second when the Track16 has successfully achieved lockup to SMPTE time code and SMPTE frame locations are being read.

#### Clock

The Clock light glows continuously when the Track16 has successfully achieved lockup to an external time base, such as SMPTE time code or the optical input.

#### Address

The Address light glows continuously when the Track16 has successfully achieved lockup to SMPTE time code.

#### Freewheel

The Freewheel light illuminates when the Track16 is freewheeling address (time code), clock or both. For details about Freewheeling, see “Freewheel Address” and “Freewheel clock” below.

#### SMPTE source

Choose the line input that is connected to the time code source. This is the input that the Track16 “listens” to for time code.

### Freewheel Address

Freewheeling occurs when there is a glitch or drop-out in the incoming time code for some reason. The Track16 can freewheel past the drop-out and then resume lockup again as soon as it receives readable time code. Choose the amount of time you would like the Track16 to freewheel before it gives up and stops altogether.

The Track16 cannot freewheel address without clock. Therefore, the *Freewheel Address* setting will always be lower than or equal to the *Freewheel Clock* setting, and both menus will update as needed, depending on what you choose.

Keep in mind that freewheeling causes the system to keep going for as long as the duration you choose from this menu, even when you stop time code intentionally. Therefore, if you are starting and stopping time code frequently (such as from the transports of a video deck), shorter freewheel times are better. On the other hand, if you are doing a one-pass transfer from tape that has bad time code, longer freewheel times will help you get past the problems in the time code.

#### The ‘Infinite’ freewheel setting

The *Infinite* freewheel setting in the *Freewheel Address* menu causes the Track16 to freewheel indefinitely, until it receives readable time code again. To make it stop, click the *Stop Freewheeling* button.

### Freewheel clock

Freewheeling occurs when there is glitch or drop-out in the incoming SMPTE time code for some reason. The Track16 can freewheel past the drop-out and then resume lockup again as soon as it receives a stable, readable clock signal.

The Track16 cannot freewheel address without clock. Therefore, the *Freewheel Address* setting will always be lower than or equal to the *Freewheel Clock* setting, and both menus will update as needed, depending on what you choose.

### The ‘Infinite’ freewheel setting

The *Infinite* freewheel setting in the *Freewheel Clock* menu causes the Track16 to freewheel indefinitely, until it receives readable time code again. To make it stop, click the *Stop Freewheeling* button.

### Stop Freewheeling

The Stop Freewheeling button stops the system if it is currently freewheeling.

## GENERATOR SECTION

The Generator section (on the right-hand side of the window in Figure 10-1) provides settings for generating SMPTE time code.

### Level

Turn the level knob to adjust the volume of the SMPTE time code being generated by the Track16. The level knob disappears when the Destination is set to *None*.

### Tach light

The Tach light blinks once per second when the Track16 is generating SMPTE time code.

### Destination

In the *Destination* menu, choose either *SMPTE* (to generate time code) or *None* (to turn it off).

### Stripe

Click this button to start or stop time code. To set the start time, click directly on the SMPTE time code display in the Generator section and type in the desired start time. Or drag vertically on the numbers.

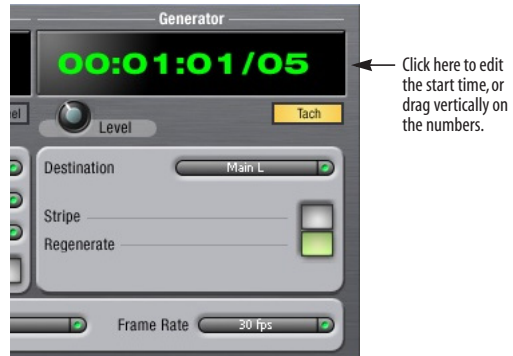


Figure 10-2: Setting the time code start time.

### Regenerate

This option, when enabled, causes the generator to generate time code whenever the Track16 is receiving SMPTE time code.

## RESOLVING CUBASE OR NUENDO TO TIME CODE

To resolve your Track16 and Cubase or Nuendo directly to SMPTE time code with no additional synchronization devices, use the setup shown in “Setup for SMPTE time code sync” on page 104. Make sure the *Master Clock Source* setting in the MOTU Audio Console window is set to *SMPTE*. Also, make sure that you’ve connected an LTC input signal to an Track16 analog input, and that you’ve specified that input in the *SMPTE Source* menu in SMPTE Console.

## SETUP FOR SMPTE TIME CODE SYNC

Here is the basic setup to resolve the Track16 system directly to SMPTE time code. The Track16 can also generate time code, under its own clock or while slaving to time code. Therefore, the Track16 can act both as an audio interface and as a SMPTE time code synchronizer to which you can slave other devices. You can use the Track16 to slave your audio software to SMPTE as well, via sample-

accurate sync (if your host software supports it). To set up your host software, refer to the following section that applies to you.

### Use this setup if you have:

- ✓ A SMPTE time code source, such as a multitrack tape deck.
- ✓ An Track16 by itself, OR with another slaved device (such as a digital mixer).
- ✓ Host software that supports sample-accurate sync.

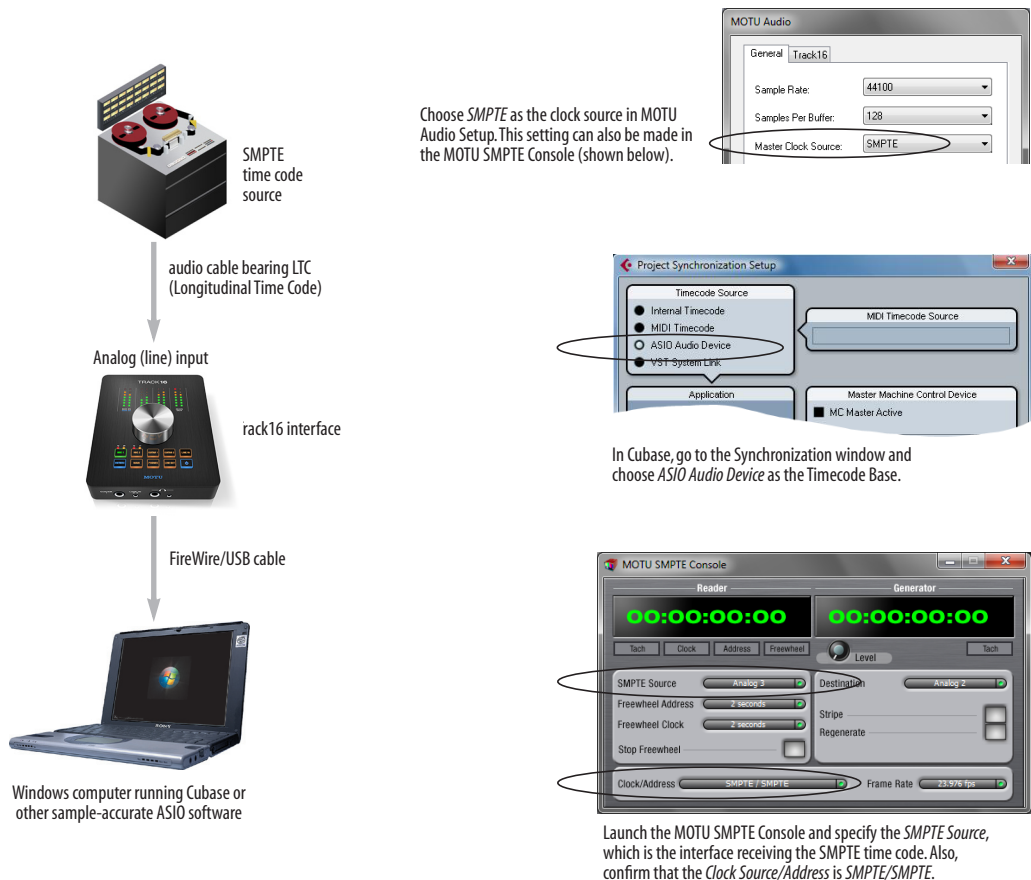


Figure 10-3: Connections for synchronizing the Track16 directly to SMPTE time code.



*Part 3*

*Appendices*



## APPENDIX A Audio I/O reference

### OVERVIEW

The MOTU Audio drivers supply text string labels for the Track16's audio inputs and outputs to clearly identify each one, but some applications do not display these labels.

The following sections show how you can identify each input and output in a numbered list like this.

### Inputs at 1x sample rates

Inputs are always listed in the same order as follows, when operating the Track16 at 1x sample rates (44.1 or 48 kHz):

Input 44.1/48 kHz	Channels	List position	Comment
Mic	2	1-2	-
Guitar	2	3-4	-
Line	4	5-8	-
Stereo return	2	9-10	See "Return Assign" on page 42.
Reverb return	2	11-12	See "Reverb return" on page 42.
Optical	8 ADAT 2 TOSLINK	13-20 13-14	-

### Inputs at 2x sample rates

When operating the Track16 at a 2x sample rate (88.2 or 96 kHz), inputs are listed as follows:

Input 88.2/96 kHz	Channels	List position	Comment
Mic	2	1-2	-
Guitar	2	3-4	-
Line	4	5-8	-
Stereo return	2	9-10	See "Return Assign" on page 42.
Reverb return	2	11-12	See "Reverb return" on page 42.
Optical	4 ADAT 2 TOSLINK	13-16 13-14	-

### Inputs at 4x sample rates

When operating the Track16 at a 4x sample rate (176.4 or 192 kHz), inputs are listed as follows:

Input 176.4/192 kHz	Channels	List position	Comment
Mic	2	1-2	-
Guitar	2	3-4	-
Line	4	5-8	-
Stereo return	2	9-10	See "Return Assign" on page 42.
Reverb return	Not available	-	-
Optical	Not available	-	-

### Outputs at 1x sample rates

Outputs are always listed in the same order as follows, when operating the Track16 at 1x sample rates (44.1 or 48 kHz):

<b>Output</b>	<b>Channels</b>	<b>List position</b>	<b>Comment</b>
Main out	2	1-2	-
Phones	2	3-4	-
Line	2	5-6	-
Optical	8 ADAT 2 TOSLINK	7-14 7-8	-

### Outputs at 2x sample rates

When operating the Track16 at a 2x sample rate (88.2 or 96 kHz), outputs are listed as follows:

<b>Output</b>	<b>Channels</b>	<b>List position</b>	<b>Comment</b>
Main out	2	1-2	-
Phones	2	3-4	-
Line	2	5-6	-
Optical	8 ADAT 2 TOSLINK	7-10 7-8	-

### Outputs at 4x sample rates

When operating the Track16 at a 4x sample rate (176.4 or 192 kHz), outputs are listed as follows:

<b>Output</b>	<b>Channels</b>	<b>List position</b>	<b>Comment</b>
Main out	2	1-2	-
Phones	2	3-4	-
Line	2	5-6	-
Optical	Not available	-	-

## APPENDIX B Troubleshooting

### *When I press the METERS button, all the other buttons change color. What's going on?*

You have accidentally put the Track16 in Lockout mode, which lets you change the color scheme for the button LEDs. To exit this mode, hold down the METERS button for 2 seconds. To learn more about this mode, and how you can specify the color scheme you prefer, see “Lockout mode” and “Button LED color schemes” on page 40.

### *Why does the start sound not play through the Track16?*

The Track16 will only playback audio at a sampling rate of 44.1, 48, 88.2 or 96 kHz.

### *The computer freezes when it starts up*

If the computer is unable to boot up, it may be a conflict with the WDM Driver. As soon as the computer boots up, Windows will try and initialize the WDM Driver. If this fails, your computer will hang. To determine if the WDM Driver is the problem, boot up in Safe Mode or remove the MOTU Audio software with the Add/Remove Programs Control Panel then restart. If that fixes the problem, visit [motu.com](http://motu.com) to download the very latest drivers and then install them.

### *'New hardware detected' window*

If you connect your Track16 before running the MOTU Audio installer, Windows will prompt you that new hardware has been detected. Cancel this, and run the Track16 Software Installer, rather than allowing Windows to locate the drivers.

### *Clicks and pops under word clock sync*

Many problems result from incorrect word clocking. It is essential that all digital devices in the system be word locked. Consult “Choosing a clock source for optical connections” on page 25 for detailed information on how to sync with other

gear that is connected optically. Whenever there is any weird noise or distortion, suspect incorrect digital clocking.

### *Clicks and pops due to hard drive problems*

If you have checked your clock settings and you are still getting clicks and pops in your audio, you may have a drive related problem. Set your Master Clock Source to *Internal* and try recording just using the analog inputs and outputs of the Track16. If you encounter the same artifacts you may want try using another drive in your computer. Clicks and pops can also occur when the drive is severely fragmented or there are other drive-related issues. If you are using a FireWire drive on the same bus as the Track16, it could be that the FireWire bus is overloaded (too many devices on the same bus). Try removing all devices except the Track16.

### *Connecting or powering gear during operation*

It is not recommended that you connect/disconnect, or power on/off devices connected to the Track16 while recording or playing back audio. Doing so may cause a brief glitch in the audio.

### *Track16 inputs and outputs are not available in host audio software*

Make sure that the inputs and outputs are enabled. See “Working with Track16 inputs and outputs” on page 46.

### *No optical inputs or outputs are available in host audio application*

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

### *Monitoring - How to monitor inputs?*

Please refer to the documentation for the audio application that you are using. If your application does not support input monitoring, you will need

to use the Track16's hardware-based CueMix FX monitoring feature. Please see chapter 8, "Reducing Monitoring Latency" (page 49).

### *Controlling monitoring latency*

See chapter 8, "Reducing Monitoring Latency" (page 49).

## **CUSTOMER SUPPORT**

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

## **REPLACING DISCS**

If your installer disc becomes damaged, our Customer Support Department will be glad to replace it. You can request a replacement disc by calling our business office at (617) 576-2760 and asking for the customer service department. In the meantime, you can download the latest drivers from [www.motu.com](http://www.motu.com).

## **TECHNICAL SUPPORT**

If you are unable, with your dealer's help, to solve problems you encounter with the Track16 system, you may contact our technical support department in one of the following ways:

- Tech support hotline: (617) 576-3066 (Monday through Friday, 9 a.m. to 6 p.m. EST)

- Online support: [www.motu.com/support](http://www.motu.com/support)

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

- The serial number of the Track16 system. This is printed on a label placed on the bottom of the Track16 rack unit. You must be able to supply this number to receive technical support.
- A brief explanation of the problem, including the exact sequence of actions which cause it, and the contents of any error messages which appear on the screen.
- The pages in the manual which refer to the parts of the Track16 or AudioDesk with which you are having trouble.
- The version of your computer's operating system.

We're not able to solve every problem immediately, but a quick call to us may yield a suggestion for a problem which you might otherwise spend hours trying to track down.

If you have features or ideas you would like to see implemented, we'd like to hear from you. Please write to the Track16 Development Team, MOTU Inc., 1280 Massachusetts Avenue, Cambridge, MA 02138 or send an email to: [customerservice@motu.com](mailto:customerservice@motu.com).

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