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Part 1
Getting Started
Quick Reference: Track16 Operation

These 7-segment level meters have a range from -42 dB to -1 dB. The top red LED illuminates when the signal reaches 1 dB below full scale — for even just one sample.

Push the METERS button below to toggle between the white labels and blue labels.

Level meters for the two mic inputs or the eighth-inch stereo mini jack on the front.

Level meters for the two guitar inputs or the phone jacks on the front.

Level meters for the line inputs or the line outputs.

Dedicated level meters for the main outs.

To toggle the -20dB pad (P) for a mic input, push the MIC 1 or MIC 2 button and then push the knob.

To toggle 48V phantom power (V), push and hold the knob.

To control the trim level for an input, press its button to select it (the button turns green) and then use the knob to adjust input level.

Push the METERS button to toggle the meters between the channels labeled in white and those labeled in blue.

To control the volume for an output, press its button to select it (the button turns green) and then use the knob to adjust level.

This multi-function knob controls the volume, trim, or bus level of the currently selected input or output channel (or pair). Use the buttons below to select the desired input or output. When an output is selected, push the knob to mute or unmute it. You can do the same when controlling an input’s CueMix bus volume (as explained below).

To control an input’s CueMix bus volume (input fader), hold the desired output bus (MAIN, PHONES, or LINE OUT) while pressing the desired input here so that both buttons turn green. Then use the knob to adjust the CueMix input level. Push to mute/unmute.
Quick Reference: Track16 Connectors

1. This is one of two hi-Z guitar inputs. The second guitar input is on the front panel (1).
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 Setup software. (See "Optical input/output" on page 38 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. 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 38. 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 Setup

The Track16 driver provides a stereo return back to the computer. This return feeds the signal on any Track16 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 Track16 only, back into the computer.

Determines the clock source for your Track16. 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 Track16 to other clock sources.

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.

The Track16 driver provides a stereo return back to the computer. This return feeds the signal on any Track16 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 Track16 only, back into the computer.

Click the tabs to access general MOTU interface settings or settings specific to the Track16 (or other connected interface.)

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.

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

This button opens another dialog that lets you assign your own customized names to each Track16 input and output. For example, if you have a lead vocal mic plugged into input 1, you could name it “Lead Vox”. Your customized names then appear in your host audio application (if it supports Core Audio input naming).

These foot switch settings are for other MOTU audio interfaces and don’t apply to the Track16.

In the standard Mac OS X fashion, the console appears in the dock when you launch it. If the Launch option is checked (as shown above), the icon appears as soon as you switch on your Track16 interface. If you click and hold on the dock icon (instead of clicking it) or control-click, a menu of hardware settings appears as shown to the right. You can view and configure any hardware settings from this menu, without opening the console window.
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
- AudioDesk™, full-featured audio workstation software for Mac OS X that supports both 16-bit and 24-bit recording
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:

<table>
<thead>
<tr>
<th>Connection</th>
<th>Input</th>
<th>Output</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mic 24-bit 192 kHz on XLR with preamp</td>
<td>2</td>
<td></td>
</tr>
<tr>
<td>Guitar 24-bit 192 kHz on hi-Z quarter-inch</td>
<td>2</td>
<td></td>
</tr>
<tr>
<td>Analog 24-bit 192 kHz on bal/unbal TRS</td>
<td>2</td>
<td>4</td>
</tr>
<tr>
<td>Analog 24-bit 192 kHz on unbal stereo &quot;mini&quot;</td>
<td>2</td>
<td>0</td>
</tr>
<tr>
<td>Headphone output*</td>
<td>-</td>
<td>4</td>
</tr>
<tr>
<td>ADAT optical digital†</td>
<td>8</td>
<td>8</td>
</tr>
<tr>
<td><strong>Total</strong></td>
<td>16</td>
<td>16</td>
</tr>
</tbody>
</table>

* 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 38 for details about optical bank operation.

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

**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 independently, 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 T rack16's standard MIDI IN and MIDI OUT jacks supply 16 channels of MIDI I/O to and from the computer. Timing accuracy can be sample-accurate with host software that supports it.

On-board SMPTE synchronization
The T rack16 can resolve directly to SMPTE time code via any analog line input, without a separate synchronizer. The T rack16 can also generate SMPTE time code via any analog output. The T rack16 provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy.

The included MOTU SMPTE Setup™ software 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 them.

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 28.

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 Setup
MOTU Audio Setup 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 Setup
MOTU SMPTE Setup 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.

**AUDIODESK**
AudioDesk is a full-featured, 24-bit audio workstation software package included with the Track16 system (for Mac OS X only). AudioDesk provides multi-channel waveform editing, automated virtual mixing, graphic editing of ramp automation, real-time effects plug-ins with 32-bit floating point processing, crossfades, support for many third-party audio plug-ins, background processing of file-based operations, sample-accurate editing and placement of audio, and more.

**DIGITAL PERFORMER**
The Track16 system is fully integrated with MOTU’s award-winning Digital Performer audio sequencer software package.

**OTHER HOST AUDIO SOFTWARE**
The Track16 system includes a standard Mac OS X Core Audio driver for multichannel I/O with any audio application that supports Core Audio.
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

MAC SYSTEM REQUIREMENTS
The Track16 system requires the following Mac system:

■ PowerPC G4 CPU 1 GHz or faster (including PowerPC G5 CPUs and all Intel processor Macs)
■ 1 GB RAM; 2 GB or more recommended
■ Mac OS X version 10.5, 10.6, or 10.7; v10.5.8 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
OR
■ Fill out and mail the included product registration card

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

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

Thank you for taking the time to register your new MOTU products!
CHAPTER 3  Installing the 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.

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 MOTU Audio Installer application.

4  Follow the directions that the installer gives you.

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

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

<table>
<thead>
<tr>
<th>Software component</th>
<th>Purpose</th>
<th>For more information</th>
</tr>
</thead>
<tbody>
<tr>
<td>MOTU Audio and MIDI drivers</td>
<td>Provides multi-channel audio input and output for MOTU FireWire and USB Audio devices with host audio software.</td>
<td>“MOTU Audio driver” on page 19</td>
</tr>
<tr>
<td>MOTU Audio Setup</td>
<td>Provides access to all of the settings in the Track16 and other MOTU interfaces. Required for Track16 operation.</td>
<td>chapter 5, “MOTU Audio Setup” (page 35)</td>
</tr>
<tr>
<td>CueMix FX</td>
<td>Gives you complete control over the Track16’s CueMix FX onboard mixer, which provides no-latency monitoring, mixing and processing of live inputs through your Track16.</td>
<td>chapter 9, “CueMix FX” (page 57)</td>
</tr>
<tr>
<td>MOTU SMPTE Setup</td>
<td>Provides access to the Track16 system’s SMPTE time code sync features.</td>
<td>chapter 10, “MOTU SMPTE Setup” (page 103)</td>
</tr>
<tr>
<td>AudioDesk</td>
<td>Provides complete multi-track recording, mixing and processing. Optional.</td>
<td>AudioDesk User Guide</td>
</tr>
</tbody>
</table>
For details about using the Track16 with Core Audio applications, see chapter 7, “Configuring Host Audio Software” (page 43).

MOTU AUDIO SETUP
MOTU Audio Setup (available in the Applications folder) 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 Setup” (page 35).

CORE MIDI AND AUDIO MIDI SETUP
Core MIDI is the “under-the-hood” portion of Mac OS X that handles MIDI services for MIDI hardware and software. Core MIDI provides many universal MIDI system management features, including MIDI communication between your Track16 interface and all Core MIDI compatible software.

Audio MIDI Setup is a utility included with Mac OS X that allows you to configure your Track16 interface for use with all Core MIDI compatible applications. Audio MIDI Setup provides:

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

Launching Audio MIDI Setup
1 Make sure your Track16 interface is connected and turned on.
2 Launch the Audio MIDI Setup utility.

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

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

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

Connecting MIDI devices to the Track16
Once your Track16 interface appears in Audio MIDI Setup, you are ready to add devices, indicate how they are connected, and identify properties they may have for particular purposes. This information is shared with all Core MIDI compatible applications.

To add a device in Audio MIDI Setup:
1 Click Add Device.
2 Drag on its input and output arrows to draw connections to the Track16 that match its physical connection.

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

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

5 When you are finished, quit Audio MIDI Setup. Your configuration is automatically saved as the default configuration, and it is shared with all Core MIDI-compatible software.

**CUEMIX FX**

CueMix FX (available in the Applications folder) provides control over the 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 57).

**MOTU SMPTE SETUP**

MOTU SMPTE Setup (available in the Applications folder) 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 Setup” (page 103).
AUDIODESK WORKSTATION SOFTWARE

AudioDesk is an advanced workstation software package for the Track16 that lets you record, edit, mix, process, bounce and master multi-track digital audio recording projects. Advanced features include real-time 32-bit effects processing, 24-bit recording, and much more.

See the AudioDesk User Guide included with your Track16 system for details.

Figure 3-5: AudioDesk for Mac OS X.
CHAPTER 4 Installing the Track16 Hardware

OVERVIEW
Here's an overview for installing the Track16:

Connect the Track16 interface ......................... 23
Connect the Track16 to the computer.

Connect audio inputs and outputs ..................... 25
Make optical and analog connections as desired.

Connect MIDI gear ........................................ 28
Connect a controller, synth or control surface.

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Choose from among several convenient options.

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An example setup for computer-based mixing/FX.

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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 28.

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 In the Apple menu, choose About this Mac.
2 Click the More Info button to open System Profiler.
3 In the Contents pane, select USB.
4 Look at the devices in the USB Device Tree. A device named USB High-Speed Bus represents a USB 2.0 root hub. A device named USB Bus represents a USB 1.1 root hub.

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:

<table>
<thead>
<tr>
<th>To switch to</th>
<th>Do this</th>
</tr>
</thead>
<tbody>
<tr>
<td>FireWire</td>
<td>Hold down the MIC 1 button while powering up</td>
</tr>
<tr>
<td>USB</td>
<td>Hold down the MIC 2 button while powering up</td>
</tr>
</tbody>
</table>

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.
5 The red “V” LED illuminates (Figure 4-2).

6 To disable phantom power, simply repeat this procedure. The red “V” LED turns off.

The red “V” LED illuminates (Figure 4-2).

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 63.

-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 42.

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 63. 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:

<table>
<thead>
<tr>
<th>EBU-R68 measurement</th>
<th>dBV</th>
<th>dBu</th>
<th>dBFS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Meter alignment</td>
<td>0</td>
<td>0</td>
<td>-18</td>
</tr>
<tr>
<td>Peak level</td>
<td>+8</td>
<td>+8</td>
<td>-10</td>
</tr>
<tr>
<td>Headroom</td>
<td>10 dB typical peak to clip</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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 +12 dB of digital trim (boost) for the optical input, which can be adjusted from CueMix FX (“Input trim” on page 63).

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:

<table>
<thead>
<tr>
<th>Format</th>
<th>44.1 or 48 kHz</th>
<th>88.2 or 96 kHz</th>
</tr>
</thead>
<tbody>
<tr>
<td>ADAT optical</td>
<td>8 channels</td>
<td>4 channels of SMUX</td>
</tr>
<tr>
<td>TOSLink</td>
<td>stereo</td>
<td>stereo</td>
</tr>
</tbody>
</table>

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 Setup" (page 35).

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.](image)

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 ports

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.](image)

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:
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.

<table>
<thead>
<tr>
<th>Voltage</th>
<th>Amperage</th>
</tr>
</thead>
<tbody>
<tr>
<td>9 volts</td>
<td>1.33 amps</td>
</tr>
<tr>
<td>12 volts</td>
<td>1 amp</td>
</tr>
<tr>
<td>18 volts</td>
<td>0.66 amps</td>
</tr>
</tbody>
</table>
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 four 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 Setup displays the settings for one interface at a time. To view the settings for an interface, click its tab.

Resolving to other Core audio drivers

The MOTU FireWire/USB driver can resolve to other Core Audio drivers. This allows the Track16 and other MOTU interfaces to resolve to other audio interfaces running simultaneously on the same computer, such as the Mac’s built-in audio, a PCI-424 core system, or even 3rd-party interfaces, without the need for external word clock connections between the devices. Doing so ensures that audio tracks being played or recorded by your MOTU FireWire/USB interface will not drift apart from tracks on the other device during long playback or recording passes.

To synchronize multiple devices via the MOTU FireWire/USB driver, choose one device (PCI, FireWire, USB or otherwise) as the master clock and then slave the Track16 and other devices to it. All interfaces will remain resolved to each other via the master interface.

Make the Clock Source settings for each interface as follows:

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

Operating multiple FireWire interfaces at high sample rates

Four 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 Track16
CHAPTER 5  MOTU Audio Setup

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

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Main Out Assign ............................................................. 38
Return Assign ................................................................. 38
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Enable Pedal ................................................................. 39

ACCESSING THE TRACK16 SETTINGS
There are several ways to access Track16 settings:

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

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

  - Most applications provide access only to basic settings such as sample rate and clock source. For access to all settings, open MOTU Audio Setup using one of the techniques above.
**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**
The General tab (Figure 5-1) provides settings that apply to all connected MOTU FireWire and USB Audio interfaces.

**‘TRACK16’ TAB SETTINGS**

**Enable Core Audio Volume Controls**
Under Mac OS X 10.6 (Snow Leopard) or later, enabling this option allows you to use the Mac’s volume control, such as the Output volume slider shown in Figure 5-2 on page 38, or the volume keys on your Mac’s keyboard, to control the output level of the Track16 when it is being used as the audio output hardware for your Mac. The Default Stereo Output setting (page 37) determines the stereo outputs to be controlled by the Mac’s volume controls.

**Sample Rate**
Choose the desired Sample Rate for recording and playback. The 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’ 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 Setup.

---

*Figure 5-1: MOTU Audio Setup gives you access to all of the settings in the Track16 hardware.*
Clock Source
The 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 Clock Source menu when the optical input bank is enabled and set to the ADAT Optical format, as explained in “Optical input/output” on page 38.

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 27.

SMPTE
Choose this setting to resolve the Track16 directly to SMPTE time code (L TC) being received via one of the Track16’s audio inputs. For details, see “Setup for SMPTE time code sync” on page 106 and chapter 10, “MOTU SMPTE Setup” (page 103).

Built-in Audio
Choose this setting to resolve the Track16 to your Mac’s built-in audio. Doing so will ensure that audio streams playing back from or recorded by the Track16 will not drift apart from audio streams simultaneously played or recorded by the Mac’s built-in mic, speakers or audio output.

Other audio devices (drivers)
The MOTU FireWire Audio Driver has the ability to resolve to other Core Audio drivers. Doing so will ensure that audio streams playing back from or recorded by the Track16 will not drift apart from audio streams simultaneously played or recorded by the other devices.

Default Stereo Input/Output
In the System Preferences window (Figure 5-2), Mac OS X lets you choose third-party hardware such as the Track16 for your Mac sound input and output. The system input and output can be used for alert sounds and general audio I/O for applications like iTunes, iMovie, etc.
The Mac OS X sound preferences let you use the Track16 for general stereo audio input and output for your Mac.

The Default Stereo Input and Default Stereo Output settings in MOTU Audio Setup (Figure 5-1 on page 36) let you specify the stereo input and output on the Track16 to be used when it is chosen as the audio I/O device in the system preferences.

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

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

Optical input/output
The Optical input and Optical output settings let you choose between 8-channel ADAT optical ('lightpipe') and stereo S/PDIF optical ('TOSLINK') as the format for the 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).

‘GENERAL’ TAB SETTINGS

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

Edit Channel Names
Click the *Edit Channel Names* button to open the Channel Names window (Figure 5-3). This window lets you edit the names of the Track16 inputs and outputs, as they appear in your host audio software. For example, when you click on a menu that displays the Track16 inputs (or outputs), you will see the names you specify in this window (e.g. “vocal mic”, “lead guitar”, etc.), instead of the default generic names (“Analog 1”, “Analog 2”, etc.).

Not all Mac OS X audio software supports channel names. If not, you’ll see generic port names in your host audio software. See Appendix A, “Audio I/O reference” page (111).

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

<table>
<thead>
<tr>
<th>Button</th>
<th>LED color</th>
</tr>
</thead>
<tbody>
<tr>
<td>METERS</td>
<td>Off</td>
</tr>
<tr>
<td>POWER</td>
<td>Blue</td>
</tr>
<tr>
<td>Currently selected input or output</td>
<td>Green</td>
</tr>
<tr>
<td>Other inputs and outputs</td>
<td>Amber</td>
</tr>
<tr>
<td>Phantom power (V) and Pad (P) LEDs</td>
<td>Off</td>
</tr>
</tbody>
</table>

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.
Configuring Host Audio Software

OVERVIEW
The Track16 provides multi-channel audio input and output for Core Audio-compatible audio applications, including MOTU's Digital Performer and AudioDesk, Apple's Logic Pro, Logic Express, SoundTrack Pro, and GarageBand, and other third-party software applications such as Ableton Live, Avid Pro Tools, Cockos Reaper, Propellerhead Reason and Record, Steinberg Cubase and Nuendo, and others.

AudioDesk is included with your Track16 system. For complete information about all of AudioDesk's powerful workstation features, refer to the AudioDesk User Guide included with your Track16 system.

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

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

- chapter 3, “Installing the Track16 Software” (page 19)
- chapter 4, “Installing the Track16 Hardware” (page 23).

RUN MOTU AUDIO SETUP
Before you run your host audio software, launch MOTU Audio Setup to configure your Track16 hardware.

MOTU Audio Setup provides settings to configure your audio interface and enable the desired inputs and outputs. Only enabled inputs and outputs will be available to your software, so this is an important step. For complete details see chapter 5, “MOTU Audio Setup” (page 35).

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**Run MOTU Audio Setup** ............................................. 43
**Choosing the MOTU Audio driver** ............................... 44
**Reducing latency** ....................................................... 47
**Working with Track16 inputs and outputs** ................. 47
**Processing live inputs with host plug-ins** .................. 48
**Working with CueMix FX mixing and effects** ............. 48
**Synchronization** ....................................................... 49

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.

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 27.

If you are slaving the Track16 and your host software to SMPTE time code, follow the directions in chapter 10, “MOTU SMPTE Setup” (page 103).

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 Setup to choose which output pair you would like to hear on this return. This can be used, for example, to record back a final stereo mix that includes effects processing from the 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 MOTU Audio driver from this menu.
Digital Performer and AudioDesk
In MOTU Digital Performer or AudioDesk, choose Configure Audio System > Configure Hardware Driver from the Setup menu, as shown in Figure 7-2. Choose the MOTU Track16 from the list of drivers.

![Figure 7-2: Enabling the Track16 in Digital Performer](image)

Pro Tools
In Avid Pro Tools, go to the Setup menu and choose Playback Engine as shown in Figure 7-3. Choose the MOTU Track16 from the Current Engine menu. For information about the H/W Buffer Size setting, see “Adjusting the audio I/O buffer” on page 53.

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

![Figure 7-3: Enabling the Track16 in Pro Tools](image)
Logic Pro and Logic Express
In Logic Pro and Logic Express, go to the Preferences window, click the Audio tab, click the Devices tab and click the Core Audio tab as shown in Figure 7-4. Choose the MOTU Track16 from the Output and Input device menus.

Soundtrack Pro
In Soundtrack Pro, access the preferences window, click the Recording tab and choose MOTU Track16 from the Input and Monitor menu as shown below in Figure 7-5.

Garage Band
In Garage Band, go to the Audio/MIDI preferences and choose MOTU Track16 from the Audio Output and Audio Input menus as shown below in Figure 7-6.

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 Track16 from the ASIO Driver menu as shown below in Figure 7-7. Activate the inputs and outputs within Cubase or Nuendo as usual.

Figure 7-4: Enabling the Track16 in Logic

Figure 7-5: Enabling the Track16 in Soundtrack Pro

Figure 7-6: Enabling the Track16 in Garage Band

Figure 7-7: Enabling the Track16 in Cubase
**CONFIGURING HOST AUDIO SOFTWARE**

**Live**
In Ableton Live, access the preferences window and click the Audio tab. Choose Core Audio from the Driver Type menu. Choose the MOTU Track16 from the Input Audio Device and Output Audio Device menus as shown below in Figure 7-8. To enable or disable input or output channels, click the Input Config or Output Config buttons.

**Reason and Record**
In Propellerhead Reason or Record, go to the Preferences window, choose Audio preferences from the menu and choose MOTU Track16 from the Audio Output menu as shown below in Figure 7-9.

**Reaper**
In Cockos Reaper, access the Preferences and click Devices under the Audio preferences. Choose MOTU Track16 from the Audio Device menu as shown below in Figure 7-10.

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

**REDUCING LATENCY**
On Mac OS X, audio I/O buffer size is handled by the host audio application rather than the Track16 Core Audio driver.

Most audio software applications provide an adjustable audio buffer setting that lets you control the amount of delay you’ll hear when monitoring live inputs through your host audio software or processing them with software plug-ins.

For information, see the “Adjusting the audio I/O buffer” section of chapter 8, “Reducing Monitoring Latency” (page 51).

**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 Setup to make sure they are enabled. If you don’t plan to use either optical bank, turn it off to conserve computer bandwidth.

Remember that the main outs and headphone output can be configured in MOTU Audio Setup to mirror other outputs. In addition, the 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 38, “Phones Assign” on page 38 and “Main Out Assign” on page 38.

**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 81.

**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 38.

**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 (111).

**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 51).

**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 57) 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 27. 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 Core Audio’s sample-accurate positioning protocol, or if it can slave to MIDI Time Code, 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 106.
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 55.

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.
MONITORING LIVE INPUT

There are two ways to monitor live audio input with a 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.

Under Mac OS X, audio I/O buffer size is handled by the host audio application (not the Track16 Core Audio driver). Most audio software applications provide an adjustable audio buffer setting that lets you control the amount of delay you'll hear when monitoring live inputs or processing them with software plug-ins. Below are a few examples.

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

1. Live input (from mic, guitar, etc.) enters the MOTU interface.
2. CueMix FX immediately patches the live mic signal directly to the main outs (or other output), completely bypassing the computer. This signal could be dry, or with Track16 effects processing, such as EQ, compression or Classic Reverb.
3. Mic signal is mixed with the main outs, and you can control the volume (relative to the rest of the mix) with the mic's fader in CueMix FX.

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.
REDUCING MONITORING LATENCY

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Figure 8-4: In Cubase or Nuendo, choose Devices menu > Device Setup. Select MOTU Track16, then click the Control Panel button to access the window above and the Buffer Size setting.

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

Lower latency versus higher CPU overhead
The buffer setting has a large impact on the following things:

- Patch thru latency
- The load on your computer’s CPU
- Possible distortion at the smallest settings

- How responsive the transport controls are in AudioDesk, Digital Performer or other audio software

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

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

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

Transport responsiveness
Buffer size also impacts how quickly your audio software will respond when you begin playback, although not by amounts that are very noticeable. Lowering the buffer size will make your software respond faster; raising the buffer size will make it a little bit slower, but barely enough to notice.
Effects processing and automated mixing
Reducing latency with the buffer size setting has another benefit: it lets you route live inputs through the real-time effects processing and mix automation of your audio software.

CUEMIX FX HARDWARE MONITORING
The 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 57).

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

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

- Digital Performer
- AudioDesk

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

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

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

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

![Input Monitoring Mode](image)

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

Once enabled, CueMix FX monitoring is tied with Digital Performer or AudioDesk’s *Audio Patch Thru* feature: when you record-enable a track, the track’s input is routed directly to its output (via CueMix FX in the Track16 hardware). For example, if you record-enable a track called *guitar* in your DP or AudioDesk project, and its audio input assignment is *Analog in 2*, and its audio output assignment is optical channels 7-8, CueMix FX no-latency hardware monitoring will automatically be set up from analog in 2 to optical outputs 7-8.

Controlling CueMix FX from within other software
Consult the manual for your software.
CHAPTER 9  CueMix FX

OVERVIEW
CueMix FX is a cross-platform software application that provides graphic, on-screen control for the T rack16'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.

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 software installer. It can be found in the Applications folder.

CUEMIX FX BASIC OPERATION

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

A separate mix bus for each output pair

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

<table>
<thead>
<tr>
<th>Mix name</th>
<th>Physical output</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bus 1</td>
<td>Main Out 1-2</td>
</tr>
<tr>
<td>Bus 2</td>
<td>Phones 1-2</td>
</tr>
<tr>
<td>Bus 3</td>
<td>Analog 1-2</td>
</tr>
<tr>
<td>Bus 4</td>
<td>ADAT 1-2, TOSLink 1-2, or disabled. See note below.*</td>
</tr>
<tr>
<td>Bus 5</td>
<td>ADAT 3-4 or disabled.*</td>
</tr>
<tr>
<td>Bus 6</td>
<td>ADAT 5-6 or disabled.*</td>
</tr>
<tr>
<td>Bus 7</td>
<td>ADAT 7-8 or disabled.*</td>
</tr>
<tr>
<td>Bus 8</td>
<td>Mix Return 1-2</td>
</tr>
</tbody>
</table>

*If the Track16’s optical output is disabled in MOTU Audio Setup (Figure 5-1 on page 36), 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 38.

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 60). 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 62) 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 65) 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 77). 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.
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 38), 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 78.

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.

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

Input channel focus
Click the channel focus button (Figure 9-3) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 9-7 on page 66). 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 82.

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**

Click the input name at the top of the input channel strip (Figure 9-3) to edit the name. Input names are global across all mixes. This name also appears in host audio software on the computer (if the software supports channel names).

---

*Figure 9-3: The Inputs tab.*
Input channel focus
Click the channel focus button (Figure 9-3) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 9-7 on page 66).

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:

<table>
<thead>
<tr>
<th>Input</th>
<th>Trim cut</th>
<th>Trim boost</th>
<th>Trim Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mic inputs</td>
<td>0 dB</td>
<td>+60 dB</td>
<td>60 dB</td>
</tr>
<tr>
<td>Guitar inputs</td>
<td>0 dB</td>
<td>+22 dB</td>
<td>22 dB</td>
</tr>
<tr>
<td>Analog inputs</td>
<td>-96 dB</td>
<td>+22 dB</td>
<td>118 dB</td>
</tr>
<tr>
<td>ADAT optical</td>
<td>0 dB</td>
<td>+12 dB</td>
<td>12 dB</td>
</tr>
<tr>
<td>S/PDIF (TOSLINK)</td>
<td>0 dB</td>
<td>+12 dB</td>
<td>12 dB</td>
</tr>
</tbody>
</table>

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 81 and “Configurations menu” on page 99.

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 66.

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 68).

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.

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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 68.

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 Track 16’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 Track 16. This is processing that occurs at the very end of the signal flow, after everything else (host based effects, Track 16 input or bus processing, mixing, and so on). Processing is done in the digital domain, just before the signal goes analog through the D/A converter. Output tab processing is applied to the entire output mix (all signals being mixed to the output from various sources).

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

Output name
Click the output name at the top of the output channel strip (Figure 9-6) to edit the name. Output names are global and will also appear in host audio software on the computer (if the software supports channel names).
Output channel focus
Click the channel focus button (Figure 9-6) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 9-7 on page 66). 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 82.

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 63.

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 78.

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 79.
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.**

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**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.

---

**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 62). 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 61.

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 79.

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 64. 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).

The input level meter (Figure 9-9) is the same as the input meters in the Meters tab (Figure 9-22 on page 77) 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 T rack16 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.
**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:

![Diagram showing filter response display and parameter display](image)

**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:

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:

<table>
<thead>
<tr>
<th>Control</th>
<th>unit</th>
<th>range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gain</td>
<td>dB</td>
<td>-20.00 to +20.00</td>
</tr>
<tr>
<td>Frequency</td>
<td>Hertz</td>
<td>20 Hz to 20 kHz</td>
</tr>
<tr>
<td>Q</td>
<td>n/a - see note below</td>
<td>0.01 to 3.00</td>
</tr>
</tbody>
</table>

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

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

Returning to zero (or nominal frequency)
To return a knob to zero, or it’s nominal frequency, double-click it.

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

- Frequency = 1.00 kHz
- \( Q = 1 \)
- Gain = +3, +10 and +20 dB
Type I

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

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.
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

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

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:

Overshoot is also applied to high shelf boost and cut:
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 +20 dB, 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.

**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.

**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 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 T rack16. 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.

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 67.

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.

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 62)
- Mix bus output (Figure 9-2 on page 60)
- Outputs (Figure 9-6 on page 65)

Reverb returns

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

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

Split point

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

Mix

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

Output

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

Primary controls

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

Reverb Time

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

PreDelay

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

Shelf Filter

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

Early reflections

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

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

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

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

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

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 62) and click the Focus button for the input that the talkback mic is connected to. Click the Channel tab (Figure 9-8 on page 66) 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 65) 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 Talkback 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. Option-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.
Hold down the following general modifier keys as shortcuts:

<table>
<thead>
<tr>
<th>Shortcut</th>
<th>Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>Shift key</td>
<td>Applies your action to all inputs or all outputs in the mix.</td>
</tr>
<tr>
<td>Command key</td>
<td>Applies your action to the stereo input pair, even when it is currently configured as mono.</td>
</tr>
<tr>
<td>Option key</td>
<td>Applies your action to all busses.</td>
</tr>
<tr>
<td>Shift-Option</td>
<td>Applies your action to all inputs and mixes.</td>
</tr>
<tr>
<td>Double-click</td>
<td>Returns the control to its default value (pan center, unity gain, etc.)</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Shortcut</th>
<th>Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>Shift click</td>
<td>Applies EQ button change to all input or outputs.</td>
</tr>
<tr>
<td>Option-click</td>
<td>Applies EQ enable button changes to all bands in that input or output.</td>
</tr>
<tr>
<td>Shift-Option-click</td>
<td>Applies EQ enable button changes to all bands and all inputs or outputs.</td>
</tr>
</tbody>
</table>

### FILE MENU

**Saving and loading hardware presets**

The Track16 can store up to 16 presets in its onboard 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.

**Show meter in dock icon**

This CueMix FX File menu item applies to other MOTU interfaces 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 Command-C).
2. Choose the destination mix and choose Paste from the Edit menu (or press Command-V).

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

DEVICES MENU

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

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

FFT and Spectrogram display ........................................... 83
Oscilloscope ........................................................... 85
X-Y Plot ................................................................. 91
Phase Analysis ......................................................... 94
Tuner ................................................................. 98

Choosing channels for audio analysis
The audio analysis tools follow the currently focused audio input or output. (See “Channel focus and settings” on page 59.) 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 T rack16 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 68) or as a separate window (Figure 9-28 on page 83) opened from the Devices menu (Figure 9-26).

**Filter display options menu**

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

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

**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:

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: View controls*

*Figure 9-29: 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:

- **Silence**
- **Black** to **Blue** to **Green** to **Yellow** to **Orange** to **Red**

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).

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).
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).

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

**OSciloscope**
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).
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](image)

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

<table>
<thead>
<tr>
<th>View menu setting</th>
<th>What it displays</th>
</tr>
</thead>
<tbody>
<tr>
<td>Left</td>
<td>Left channel only</td>
</tr>
<tr>
<td>Right</td>
<td>Right channel only</td>
</tr>
<tr>
<td>Split screen</td>
<td>Left channel on top; right channel on the bottom</td>
</tr>
<tr>
<td>Shared</td>
<td>Left and right on top of each other; left is green, right is red</td>
</tr>
<tr>
<td>Add</td>
<td>Left and right channels’ amplitudes are added together</td>
</tr>
<tr>
<td>Subtract L–R</td>
<td>The right channel’s amplitude is subtracted from the left channel’s amplitude</td>
</tr>
</tbody>
</table>

**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 89).

**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](image)

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.

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.

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 87) provides four modes:

<table>
<thead>
<tr>
<th>Trigger modes</th>
<th>What it does</th>
</tr>
</thead>
<tbody>
<tr>
<td>None</td>
<td>The Trigger is not active; this is the default mode. The incoming audio signal will be displayed continuously as audio is received.</td>
</tr>
<tr>
<td>Auto</td>
<td>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.</td>
</tr>
<tr>
<td>Normal</td>
<td>The display updates only when the condition is met; the last trace will be displayed until the next matching event is found.</td>
</tr>
<tr>
<td>Single Sweep</td>
<td>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 87) or by pressing the spacebar.</td>
</tr>
</tbody>
</table>

Trigger indicator

The Trigger indicator (Figure 9-39 on page 87) 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:

<table>
<thead>
<tr>
<th>Color</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Green</td>
<td>When the current Trigger criteria has been met (including when the Trigger mode is None).</td>
</tr>
<tr>
<td>Yellow</td>
<td>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 86).</td>
</tr>
<tr>
<td>Red</td>
<td>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.</td>
</tr>
</tbody>
</table>

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.

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...
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 87), 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.

**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:

<table>
<thead>
<tr>
<th>Situation</th>
<th>Meter level</th>
<th>X-Y Plot graph</th>
<th>Mathematical relationship</th>
</tr>
</thead>
<tbody>
<tr>
<td>Perfect correlation</td>
<td>+1</td>
<td>Diagonal line going from lower left to upper right:</td>
<td>y = x</td>
</tr>
<tr>
<td>Zero correlation</td>
<td>0</td>
<td>No discernible pattern</td>
<td>None</td>
</tr>
<tr>
<td>Perfectly out of phase</td>
<td>-1</td>
<td>Diagonal line going from upper left to lower right:</td>
<td>y = -x</td>
</tr>
</tbody>
</table>

![Figure 9-42: X-Y Plot](image)
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.

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.

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 93.

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.

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 93) 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:

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

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 94) 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.

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.

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 82).

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 96).

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.

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.

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 95) 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 97), 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 97.

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.](image)

<table>
<thead>
<tr>
<th>Perfectly in phase</th>
<th>One-sample delay</th>
<th>Twenty-sample delay</th>
<th>Inverted</th>
</tr>
</thead>
</table>

**Polar view**

**Rectangular view**

![Figure 9-55: Two identical audio streams in the Phase Analysis.](image)
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 82).

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 98.
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:

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

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

Saving a CueMix FX configuration as a hardware preset
To save a CueMix FX configuration as a hardware preset:

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

Saving a hardware preset as a CueMix FX configuration
To save a hardware preset as a CueMix FX configuration:

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

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

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

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

Application follows control surface

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

Share surfaces with other applications

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

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

CueMix Control Surfaces

CueMix FX includes support for the following control surface products:

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

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

Enabled

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

Configure…

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

Figure 9-57: Refer to the extensive on-line help for details about configuring CueMix FX for operation with your control surface product.
Other HUI-compatible control surfaces
Any control surface that has the ability to emulate a HUI should be compatible with CueMix FX. Just add a Mackie HUI to Audio MIDI Setup and put the control surface hardware into HUI emulation mode. Consult the control surface manual for details about how put it into HUI emulation mode.

Other control surface hardware products
If you install other control surface drivers written for CueMix FX, they will appear as separate menu items at the bottom of the Control Surfaces menu, with the same sub-menu items described above.
CHAPTER 10 MOTU SMPTE Setup

OVERVIEW
The 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 AudioDesk and Digital Performer on Mac OS X. Other hosts, such as Pro Tools, can resolve to MIDI Time Code generated by the Track16.

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

CLOCK/ADDRESS
The Clock/Address menu (Figure 10-1) provides the same global Clock Source setting as in MOTU Audio Setup (“Clock Source” on page 37), 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-
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. The Track16 driver updates the frame rate setting in Digital Performer and AudioDesk for you.

**READER SECTION**

The Reader section (on the left-hand side of the window in Figure 10-1) provides settings for synchronizing the 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 a 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.

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

Generate from sequencer
This option, when enabled, causes the generator to generate time code whenever you are running AudioDesk or Digital Performer. time code begins at the time specified by the AudioDesk or Digital Performer main transport.
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) or via MIDI Time Code (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 or MTC.

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

![MOTU SMPTE Setup](image)

Launch the MOTU SMPTE Setup to specify the time code source, frame rate, and amount of freewheel. Also, confirm that the Clock Source/Address is SMPTE/SMPTE.

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

In Pro Tools:
1. Choose Peripherals from the Setup menu.
2. Click the Synchronization tab and choose the Track16 Sync Port from the MTC Reader Port menu.

![Figure 10-3: Connections for synchronizing the Track16 directly to SMPTE time code.](image)
RESOLVING DP OR AUDIODESK TO TIME CODE
To resolve Digital Performer or AudioDesk directly to time code with no additional sync devices, use the setup shown in “Setup for SMPTE time code sync” on page 106. Choose Receive Sync from the Setup menu and choose the Sample accurate option. Make sure that the Slave to External Sync command in the Studio menu is checked. Make sure the Clock Source setting in the MOTU Audio Setup window is set to SMPTE. Also, make sure that you’ve connected an LTC input signal to the Track16 time code input, and that you’ve specified that input in SMPTE Source menu in SMPTE Setup.

RESOLVING PRO TOOLS TO TIME CODE
To resolve your Pro Tools system directly to SMPTE time code with no additional synchronization devices, use the setup shown in “Setup for SMPTE time code sync” on page 106.

1 Connect an LTC signal to a Track16 line input.
2 In MOTU SMPTE Setup, go to the SMPTE Source menu and choose the line input.
3 In MOTU SMPTE Setup, go to the Clock Source menu and choose SMPTE/SMPTE as the clock source.
4 Make the other settings in MOTU SMPTE Setup as desired. Refer to their sections in this chapter for more information.
5 In your host audio software, make the necessary settings for resolving it to MIDI Time Code. Refer to your host audio software documentation for further information. The Track16 driver provides a separate MIDI Sync Port for the time code, so when you specify the MIDI port to follow, be sure to choose the Sync Port.

RESOLVING OTHER HOSTS TO TIME CODE
The Track16 has the ability to convert incoming SMPTE time code (LTC) to MIDI Time Code and send it to your host audio software, which can resolve to it. To do so, use the setup shown in “Setup for SMPTE time code sync” on page 106. Here is the basic procedure:

1 Connect an LTC signal to a Track16 line input.
2 In MOTU SMPTE Setup, go to the SMPTE Source menu and choose the line input.
3 In MOTU SMPTE Setup, go to the Clock Source menu and choose SMPTE/SMPTE as the clock source.
4 Make the other settings in SMPTE Setup as desired. Refer to their sections in this chapter for more information.
5 In your host audio software, make the necessary settings for resolving it to MIDI Time Code. Refer to your host audio software documentation for further information. The Track16 driver provides a separate MIDI Sync Port for the time code, so when you specify the MIDI port to follow, be sure to choose the Sync Port.
Part 3
Appendices
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):

<table>
<thead>
<tr>
<th>Input</th>
<th>44.1/48 kHz</th>
<th>Channels</th>
<th>List position</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mic</td>
<td>2</td>
<td>1-2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Guitar</td>
<td>2</td>
<td>3-4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Line</td>
<td>4</td>
<td>5-8</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Stereo return</td>
<td>2</td>
<td>9-10</td>
<td>See “Return Assign” on page 44.</td>
<td></td>
</tr>
<tr>
<td>Reverb return</td>
<td>2</td>
<td>11-12</td>
<td>See “Reverb return” on page 44.</td>
<td></td>
</tr>
<tr>
<td>Optical</td>
<td>8 ADAT 2 TOSLINK</td>
<td>13-20 13-14</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Input</th>
<th>88.2/96 kHz</th>
<th>Channels</th>
<th>List position</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mic</td>
<td>2</td>
<td>1-2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Guitar</td>
<td>2</td>
<td>3-4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Line</td>
<td>4</td>
<td>5-8</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Stereo return</td>
<td>2</td>
<td>9-10</td>
<td>See “Return Assign” on page 44.</td>
<td></td>
</tr>
<tr>
<td>Reverb return</td>
<td>2</td>
<td>11-12</td>
<td>See “Reverb return” on page 44.</td>
<td></td>
</tr>
<tr>
<td>Optical</td>
<td>4 ADAT 2 TOSLINK</td>
<td>13-16 13-14</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Input</th>
<th>176.4/192 kHz</th>
<th>Channels</th>
<th>List position</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mic</td>
<td>2</td>
<td>1-2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Guitar</td>
<td>2</td>
<td>3-4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Line</td>
<td>4</td>
<td>5-8</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Stereo return</td>
<td>2</td>
<td>9-10</td>
<td>See “Return Assign” on page 44.</td>
<td></td>
</tr>
<tr>
<td>Reverb return</td>
<td>Not available</td>
<td>-</td>
<td>-</td>
<td></td>
</tr>
<tr>
<td>Optical</td>
<td>Not available</td>
<td>-</td>
<td>-</td>
<td></td>
</tr>
</tbody>
</table>
## 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):

<table>
<thead>
<tr>
<th>Output</th>
<th>44.1/48 kHz Channels</th>
<th>List position</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Main out</td>
<td>2</td>
<td>1-2</td>
<td>-</td>
</tr>
<tr>
<td>Phones</td>
<td>2</td>
<td>3-4</td>
<td>-</td>
</tr>
<tr>
<td>Line</td>
<td>2</td>
<td>5-6</td>
<td>-</td>
</tr>
<tr>
<td>Optical</td>
<td>8 ADAT</td>
<td>7-14</td>
<td>-</td>
</tr>
<tr>
<td></td>
<td>2 TOSLINK</td>
<td>7-8</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Output</th>
<th>88.2/96 kHz Channels</th>
<th>List position</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Main out</td>
<td>2</td>
<td>1-2</td>
<td>-</td>
</tr>
<tr>
<td>Phones</td>
<td>2</td>
<td>3-4</td>
<td>-</td>
</tr>
<tr>
<td>Line</td>
<td>2</td>
<td>5-6</td>
<td>-</td>
</tr>
<tr>
<td>Optical</td>
<td>8 ADAT</td>
<td>7-10</td>
<td>-</td>
</tr>
<tr>
<td></td>
<td>2 TOSLINK</td>
<td>7-8</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Output</th>
<th>176.4/192 kHz Channels</th>
<th>List position</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Main out</td>
<td>2</td>
<td>1-2</td>
<td>-</td>
</tr>
<tr>
<td>Phones</td>
<td>2</td>
<td>3-4</td>
<td>-</td>
</tr>
<tr>
<td>Line</td>
<td>2</td>
<td>5-6</td>
<td>-</td>
</tr>
<tr>
<td>Optical</td>
<td>Not available</td>
<td>-</td>
<td>-</td>
</tr>
</tbody>
</table>
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 42.

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

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

Clicks and pops under word clock sync
Many problems result from incorrect word clocking. It is essential that all digital devices in the system be word locked. Consult “Choosing a clock source for optical connections” on page 27 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 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 47.

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

Monitoring - How to monitor inputs?
Please refer to the documentation for the audio application that you are using. If your application does not support input monitoring, you will need to use the Track16’s hardware-based CueMix FX monitoring feature. Please see chapter 8, “Reducing Monitoring Latency” (page 51).
Controlling monitoring latency
See chapter 8, “Reducing Monitoring Latency” (page 51).

CUSTOMER SUPPORT
We are happy to provide customer support to our registered users. If you haven’t already done so, please take a moment to register online at MOTU.com, or fill out and mail the included registration card. Doing so entitles you to technical support and notices about new products and software updates.

REPLACING DISCS
If your installer disc becomes damaged, our Customer Support Department will be glad to replace it. You can request a replacement disc by calling our business office at (617) 576-2760 and asking for the customer service department. In the meantime, you can download the latest drivers from www.motu.com.

TECHNICAL SUPPORT
If you are unable, with your dealer’s help, to solve problems you encounter with the 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

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.
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