UltraLite-mk3 Hybrid™

User Guide for Windows

MOTU

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

• Relocate or reorient the receiving antenna

• Increase the separation between the equipment and the receiver

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

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

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

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

CE

Contents

Quick Reference: MOTU Audio Console

Determines the clock source for your UltraLite-mk3. If you're just using the analog ins and outs, set this to *Internal.* The other settings are for digital transfers via S/PDIF or synchronization to time code or other audio devices.

Check this option if the audio software you are using with the UltraLite-mk3 does not support Windows WDM drivers and instead only supports legacy MME (Wave) drivers. When checked, this option makes all of the UltraLite-mk3 inputs and outputs available to legacy Wave-driver compatible audio software.
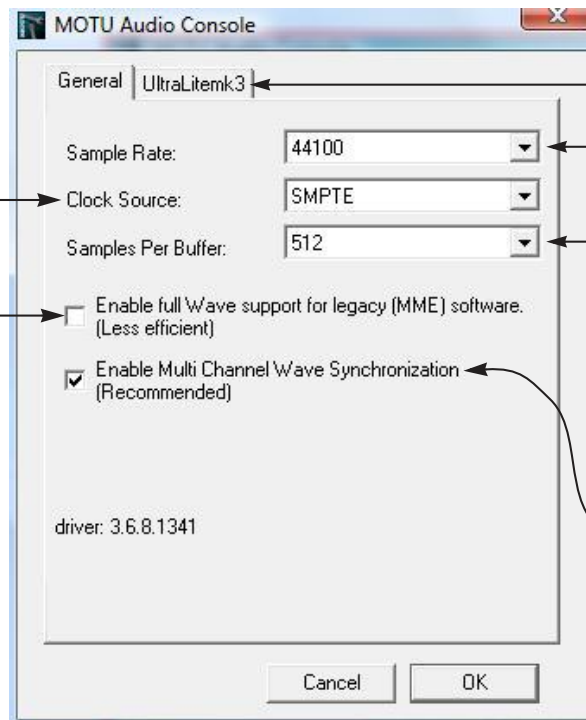
Click the tabs to access general MOTU interface settings or settings specific to the UltraLite-mk3 (or other connected interface).

Choose the global sample rate for the system here.

Choosing a smaller setting here reduces the delay you may hear when listening to live input that you are running through effects plug-ins in your software. But lower settings also increase the strain on your computer. For details, see "Samples Per Buffer" on page 31.

This option should always be left on (checked). There are only a few rare cases in which you would want to turn it off. For details, refer to the MOTU tech support database at www.motu.com.

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

Choose the output pair you would like the main outs to mirror, or choose *Main Outs* to operate them as their own independent pair.
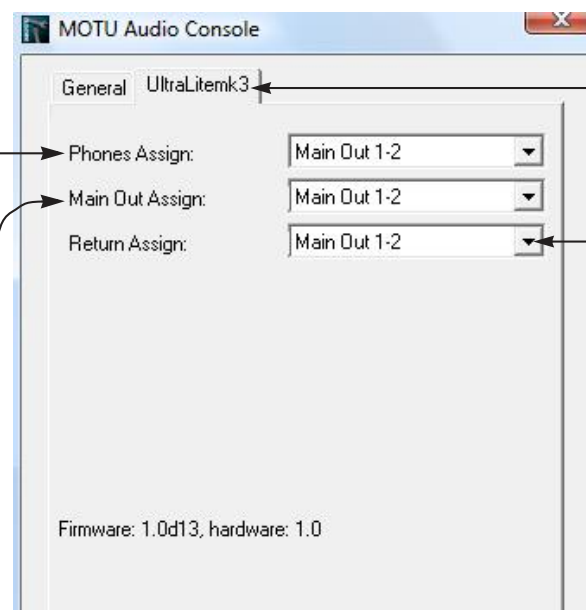
Click the UltraLite-mk3 tab to access these settings.

The UltraLite-mk3 driver provides a stereo return back to the computer. This return feeds the signal on any UltraLite-mk3 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 UltraLite-mk3 only, back into the computer.

Device Setup in Cubase

How to access these settings

There are several ways to access these settings:
- From the Windows Start menu, choose *Programs>MOTU>MOTU Audio Console.*
- From within Cubase, go to the Device Setup window, click the MOTU Audio ASIO list item and and click the *Control Panel* button.
- From within other applications, refer to their documentation.
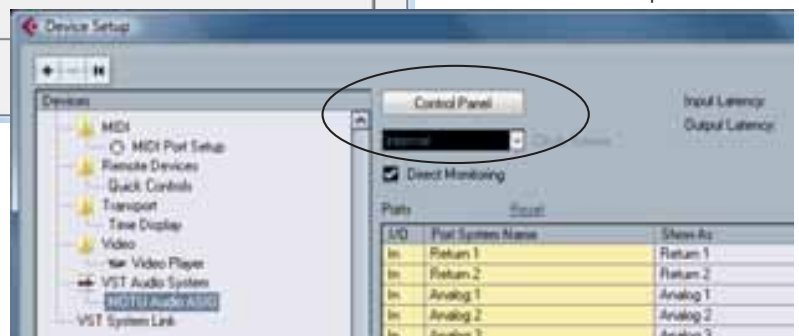
CHAPTER 1   About the UltraLite-mk3 Hybrid

OVERVIEW

The UltraLite-mk3 is a hybrid FireWire USB2 audio interface for Mac and Windows with on-board effects and mixing that offers 10 inputs and 14 outputs. 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 UltraLite-mk3 is housed in a sturdy, compact half-rack enclosure that connects directly to a computer via a standard FireWire or USB cable.

The UltraLite-mk3 offers the following main features:

■ Universal computer connectivity via FireWire or high-speed USB2

■ Bus-powered operation (FireWire only)

■ Six 24-bit analog quarter-inch (TRS) inputs

■ Ten 24-bit analog quarter-inch (TRS) outputs

■ Two combo XLR/TRS mic/guitar inputs with preamps, 48V phantom power, 20 dB pad, and Precision Digital Trim™ preamp gain adjustment

■ Operation on all analog I/O at standard sample rates up to 192 kHz

■ Digitally controlled analog trim for all analog inputs

■ RCA S/PDIF digital I/O at sample rates up to 96 kHz

■ MIDI I/O

■ On-board SMPTE synchronization

■ Headphone jack with volume control

■ Front panel volume control of the Main Outs (or any programmable combination of outputs)

■ CueMix™ FX no-latency mixing, monitoring, and effects processing

■ Classic Reverb™ with five different room types, three frequency shelves with adjustable crossover points, shelf filtering and reverb lengths up to 60 seconds.

■ Two forms of compression, including 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.

■ 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. LP and HP filters are also supplied with slopes that range from 6 to 36 dB.

■ Front-panel LCD programming for the mixer and all other settings

■ Extensive front panel metering and status LEDs

■ 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

■ CueMix FX cross-platform mixing software with attractive graphic mixing, graphic editing of parametric EQ and a convenient tabbed interface

for quick access to all features in one window. Also provides full-screen real-time FFT display, spectrogram "waterfall" display, oscilloscope, X-Y plot and linear or polar phase analysis.

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

THE ULTRALITE-MK3 REAR PANEL

The rear panel has the following connectors:

■ Ten gold-plated, balanced +4dB quarter-inch (TRS) analog outputs (with 24-bit 192 kHz converters)

■ Six gold-plated, balanced quarter-inch (TRS) analog inputs (with 24-bit 192 kHz converters)

■ One combo XLR/TRS mic/instrument input

■ RCA S/PDIF in/out

■ MIDI IN and MIDI OUT jacks

■ One 1394 FireWire A connector

■ One high-speed USB2 connector

■ DC power jack

10 inputs and 14 outputs

All UltraLite-mk3 inputs and outputs can be used simultaneously, for a total of 10 inputs and 14 outputs when operating at sample rates up to 96 kHz:

Connection	Input	Output
Analog 24-bit 192 kHz on bal/unbal TRS	6	10
Mic preamps 24-bit 192 kHz on XLR/TRS combo	2	-
SPDIF 24-bit 96kHz digital	2	2
Headphone output	-	2
Total	10	14

All inputs and outputs are discrete and can be active simultaneously.

The headphone outputs can operate as an independent output pair, or they can mirror any other UltraLite-mk3 output pair, such as the main outs.

Mic/guitar inputs with preamps

The two mic/instrument inputs (front panel and rear panel) are equipped with preamps and "combo" XLR/TRS jacks, which accept XLR microphone inputs or quarter-inch guitar/instruments inputs. Individual 48 volt phantom power and a 20 dB pad can be supplied independently to each mic input. The Precision Digital Trim™ knobs on the front panel for each mic/instrument input provide 60 dB of boost for the XLR mic input and 32 dB of boost for the TRS instrument input in precise 1 dB increments.

Analog

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

All ten analog outputs and six quarter-inch analog inputs are on balanced TRS +4dB quarter-inch jacks. All of these jacks can also accept unbalanced plugs.

Precision Digital Trim™

All of the UltraLite-mk3's analog inputs are equipped with digitally controlled analog trims that allow adjustments in 1 dB increments. The mic/guitar input trims can be adjusted using front-panel digital rotary encoders that provide feedback in the front panel LCD with 60 dB of adjustment for the mic input and 32 dB for the TRS input. All analog inputs, including six rear-panel TRS analog inputs, can be trimmed using the front panel LCD or using the UltraLite-mk3's included CueMix FX

control software for Mac and Windows. This allows you to fine-tune trim settings for synths, effects modules and a wide variety of analog inputs for optimum levels. Different trim configurations can then be saved as files on disk for instant recall.

S/PDIF
The UltraLite-mk3 rear panel provides RCA "coax" S/PDIF input and output.

MIDI I/O
The UltraLite-mk3's standard MIDI IN and MIDI OUT jacks supply 16 channels of MIDI I/O to and from the computer via the UltraLite-mk3's FireWire connection.

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

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

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

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

The UltraLite-mk3 has the ability to power itself from its FireWire connection to the computer, for convenient, mobile bus-powered operation.

Power supply
If you do not want the UltraLite-mk3 to draw power from the computer, and AC power is available, you can power the UltraLite-mk3 from any standard 10-18V, 12 watt DC power supply with any polarity (tip positive or negative).

THE ULTRALITE-MK3 FRONT PANEL

Mic/instrument input
See "Mic/guitar inputs with preamps" on page 10.

48V phantom power and pad
See "Mic/guitar inputs with preamps" on page 10.

Headphone output and main volume control
The UltraLite-mk3 front panel provides a quarter-inch stereo headphone output jack and volume knob. The LCD display provides feedback as you turn the knob. The VOL knob can be used to control the volume of the main out jacks on the rear panel. It can alternately be configured to control any combination of outputs.

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

The LCD also provides activity metering for all UltraLite-mk3 inputs and outputs.

16-BIT AND 24-BIT RECORDING

The UltraLite-mk3 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 UltraLite-mk3'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 UltraLite-mk3 inputs and outputs can be routed to the on-board CueMix FX 16-bus (8 stereo) digital mixer driven by hardware-based DSP with 32-bit floating point precision. The mixer allows you to apply no-latency effects processing to inputs, outputs or busses directly in the UltraLite-mk3 hardware, independent of the computer. Effects can even be applied when the UltraLite-mk3 is operating stand-alone (without a computer) as a complete portable 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 UltraLite-mk3'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 UltraLite-mk3's flexible effects architecture allows you to apply EQ and compression on every input and output (a total of 24 channels), with enough DSP resources for at least one band of parametric EQ and compression on every channel at 48 kHz. However, DSP resources are allocated dynamically and a DSP meter in the CueMix FX software (included) allows you to keep tabs on the UltraLite-mk3'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.

HOST AUDIO SOFTWARE

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

CHAPTER 2 Packing List and Windows System Requirements

PACKING LIST

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

- One UltraLite-mk3 I/O rack unit

- One 6-pin to 6-pin IEEE 1394 "FireWire" cable

- One USB cable

- One set of removable rack ears

- One UltraLite-mk3 Mac/Windows manual

- One AudioDesk Manual

- One cross-platform CD-ROM

- Product registration card

WINDOWS SYSTEM REQUIREMENTS

The UltraLite-mk3 system requires the following Windows system:

- A 1 GHz Pentium-based PC compatible or faster equipped with at least one FireWire port

- A Pentium III processor or faster is recommended

- At least 256 MB (megabytes) of RAM (512 MB or more is recommended)

- Windows XP or Vista

- A large hard drive (preferably at least 250 GB)

PLEASE REGISTER TODAY!

Please register your UltraLite-mk3 today. There are two ways to register.

- Visit www.motu.com/registration to register on line

OR

- Fill out and mail the included product registration card

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

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

IMPORTANT! Run the UltraLite-mk3 Software Installer First

OVERVIEW

INSTALL THE ULTRALITE-MK3 SOFTWARE FIRST!

Before you connect the UltraLite-mk3 audio interface to your computer and turn it on, insert the UltraLite-mk3 software CD and run the UltraLite-mk3 Software Installer. This ensures that all the UltraLite-mk3 components are properly installed in your system.

If Windows asks you to locate the drivers

If you've already connected the UltraLite-mk3 to your computer and switched it on, Windows probably issued an alert notifying you that the UltraLite-mk3 requires drivers, followed by another window asking you to locate the drivers on disk. If this happens:

1 Cancel the driver search.

2 Switch off the UltraLite-mk3.

3 Run the UltraLite-mk3 Software Installer as instructed in the next section.

INSTALLING THE ULTRALITE-MK3 SOFTWARE

To install the UltraLite-mk3 software, insert the MOTU Universal Audio Installer CD and follow the directions it gives you on your computer screen. The UltraLite-mk3 ships with the following software and drivers for Windows XP and Vista:

Software component	Purpose
MOTU Audio Console	Provides access to all of the settings in the UltraLite-mk3 hardware.
CueMix FX	Gives you complete control over the 896mk3's CueMix FX on-board mixer, which provides no-latency monitoring, mixing and processing of live inputs through your UltraLite-mk3.
MOTU ASIO Driver	Allows Cubase or other ASIO-compliant software to do multi-channel input and output with the UltraLite-mk3. Only required if you are using Cubase or another ASIO-dependent program.
MOTU WDM Driver	Allows any WDM-driver compatible audio software to do multichannel input and output with the UltraLite-mk3.
MOTU GSIF Driver	Allows you to use your UltraLite-mk3 with Tascam GigaStudio.
MOTU MIDI Driver	Provides MIDI input and output via the UltraLite-mk3 MIDI ports.

MOTU AUDIO CONSOLE

MOTU Audio Console (available in the Start menu) gives you access to all of the settings in the UltraLite-mk3, such as the clock source and sample rate. For complete details, see chapter 10, "CueMix FX" (page 61).
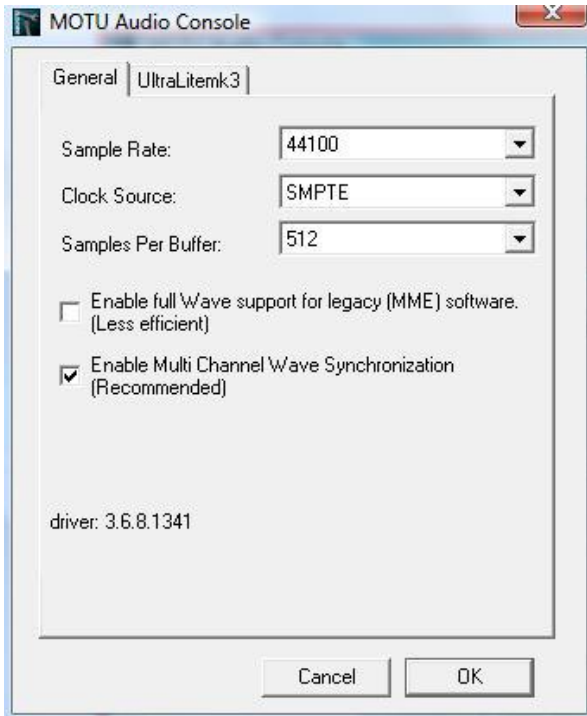

Figure 3-1: MOTU Audio Console gives you access to all of the settings in the UltraLite-mk3 hardware.

ASIO MOTU AUDIO DRIVER

ASIO stands for *Audio Streaming Input* and *Output*. The ASIO MOTU FireWire driver allows the UltraLite-mk3 to provide multi-channel input and output for Steinberg's Cubase and Nuendo software, or any other audio application that supports ASIO drivers.

The MOTU ASIO audio driver is only required if you are using Cubase (or another audio program that relies on the ASIO driver to support multi-channel I/O with the UltraLite-mk3).

The MOTU ASIO audio driver is installed by the UltraLite-mk3 Software Installer and properly registered with Windows, so you don't need to be concerned about its installation or location.

For details about using Cubase and Nuendo with the UltraLite-mk3, see chapter 7, "Cubase, Nuendo, Live and Other ASIO Software" (page 43).

THE MOTU AUDIO WDM DRIVER

The MOTU Audio WDM driver provides standard multi-channel input and output for audio applications running under Windows XP or Vista. See chapter 8, "SONAR and other WDM Software" (page 49) for details.

The MOTU Audio installer CD installs the MOTU Audio WDM driver into Windows for you.

MOTU GSIF FIREWIRE DRIVER

The MOTU GSIF Driver allows you to access the UltraLite-mk3 as an audio interface for Tascam GigaStudio and related products. The GSIF driver features low-latency multi-channel performance.

MOTU MIDI DRIVER

This driver allows you to access the UltraLite-mk3's MIDI input and output ports. The ports are published in Windows and are available to all MIDI software.

CUEMIX FX

This program provides a mixing console that gives you control over the UltraLite-mk3's no-latency CueMix FX on-board mixing and effects processing. For details, see chapter 10, "CueMix FX" (page 61).

MOTU SMPTE CONSOLE

The MOTU SMPTE Console software provides a complete set of tools for resolving the UltraLite-mk3 to SMPTE time code, and to generate SMPTE for striping, regenerating or slaving other devices to the computer. For details, see chapter 11, "MOTU SMPTE Console" (page 103).

CHAPTER 4 Installing the UltraLite-mk3 Hybrid Hardware

OVERVIEW

Here's an overview for installing the UltraLite-mk3:

IMPORTANT NOTE BEFORE YOU BEGIN!

Before you begin installing the UltraLite-mk3 (or any bus-powered device), take these important precautionary measures to avoid damaging the sensitive electrical components in your computer, the UltraLite-mk3 or other devices being connected:

■ Turn off the computer.

■ Turn off the UltraLite-mk3 (push and hold the VOL knob.

■ Turn off the power of any other devices.

■ Touch the metal casing of the UltraLite-mk3 to discharge any static electricity that you may be carrying just before the installation.

After you have made all of the necessary connections, as described in this chapter, turn on the devices in this sequence:

1. Turn on the computer.

2. Turn on the UltraLite-mk3.

3. Turn on other devices connected to the UltraLite-mk3.

CONNECT THE ULTRALITE-MK3 INTERFACE

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

Which should I use: FireWire or USB2?

If your computer does not have a FireWire port, then obviously you will need to connect the UltraLite-mk3 Hybrid to one of its hi-speed USB2 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 UltraLite can be powered solely by its FireWire connection to the computer. For complete information, see "Bus power requirements" on page 23.

If you use USB2, you must also use the DC power supply included with your UltraLite-mk3, 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 UltraLite-mk3 are switched off.

2 Plug one end of the UltraLite-mk3 FireWire cable (included) into the FireWire socket on the computer as shown below in Figure 4-1.

3 Plug the other end of the FireWire cable into the UltraLite-mk3 I/O as shown below in Figure 4-1.

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

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

If you are connecting via USB2

Some USB ports on computers are USB Version 1.1 ports, which support low-power, low-bandwidth peripherals such as your computer's mouse and keyboard. USB 1 ports do not support the high-speed operation necessary for the UltraLite-mk3. If you are not sure if the USB ports on your computer are Hi-speed USB2 or not, check your computer's documentation. Or check your computer as follows:

■ On the Mac, you can run System Profiler (*Apple menu> About This Mac> More Info...*) to view the specifications for each USB port on your Mac. Look in the hardware list for USB and click it. If the USB bus description on the right-hand side says *USB High Speed Bus*, then your Mac has USB2.

■ On Windows, open Device Manager and expand the Universal Serial Bus section. There should be an "Enhanced" USB host controller present.

☛ Due to the way all USB 1.1 and USB 2.0 devices share bandwidth, you will achieve better overall performance by placing 1.1 and 2.0 devices on separate busses, if possible. For details, refer to the tech support database at www.motu.com.

1 Before you begin, make sure your computer and the UltraLite-mk3 are switched off.

2 Plug the flat "type A" plug of the UltraLite-mk3 USB cable (included) into a USB2-equipped socket on the computer as shown below in Figure 4-1.

3 Plug the squared "type B" plug of the USB cable into the UltraLite-mk3 I/O as shown below in Figure 4-1.

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

CONNECT AUDIO INPUTS AND OUTPUTS

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

- 10 balanced quarter-inch analog outputs

- 6 balanced quarter-inch analog inputs

- 2 XLR/quarter-inch "combo" analog inputs with preamps

- 1 pair of RCA S/PDIF in and out

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

Mic/instrument inputs

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

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

Phantom power

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

Trim

The XLR mic input and the TRS instrument input are equipped with 60 dB and 32 dB of trim control, respectively. Use the digital trim encoders on the front panel to adjust the input level as needed for each input. The LCD provides visual feedback as you turn the trim knob. The UltraLite-mk3's input trims are digital controlled, so they allow you to make fine-tuned adjustments in precise 1dB increments. You can also adjust trim in the MOTU CueMix FX software. See "Input trim" on page 67.

20 dB pad

The mic input (XLR jack) is equipped with a 20 dB pad, so "hot" signals are best connected via an XLR cable so that you can use the pad. To toggle the 20 dB pad for a mic input, quickly push its TRIM rotary encoder. The green LED below will turn on or off accordingly.

Since the pad is not available on the TRS jack, hot signals connected via the TRS jack will probably overdrive the input.

Combo jack summary

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

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

Quarter-inch analog

The quarter-inch analog inputs (3-8) and outputs (1-8 plus main out) are balanced (TRS) connectors that can also accept an unbalanced plug.

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

Quarter-inch analog input trims

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

adjust these trims using CueMix FX, see "Input trim" on page 67. To adjust the trims using the front panel LCD:

1 Push the PARAM knob repeatedly until you see CUEMIX in the LCD display.

2 Push the CHANNEL knob repeatedly until you see "I:" (which stands for *Input*) in the CHANNEL section of the LCD (Figure 4-3).

3 Turn the CHANNEL KNOB until you see the desired analog input or input pair. For example, analog inputs 1-2 appear as "I:An 1-2" (Figure 4-3), which means *Input analog 1-2.*

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

Figure 4-3: The settings for analog inputs 1 and 2 (as a pair).

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

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

6 Turn the VALUE knob to adjust the trim.

Separate main outs

The main outputs operate as an independent pair (they don't share signal with any other output pair). In a standard studio configuration, the main outs are intended for a pair of studio monitors, but they can also be used as regular outputs for any purpose. Their reference level is +4dB. Main out volume is controlled by the VOL knob on the front panel: push it repeatedly until you see MASTER and then turn it to adjust the master volume output. This knob can be programmed to control any combination of outputs you wish.

S/PDIF

If you make a S/PDIF digital audio connection to another device, be sure to review the digital audio clocking issues, as explained in "Connecting and syncing S/PDIF devices" on page 22.

The UltraLite-mk3 on-board mixer also supplies 12dB of digital trim adjustment for the S/PDIF input pair, which can be adjusted from CueMix FX ("Input trim" on page 67) or the front panel (using the same procedure as explained earlier for "Quarter-inch analog input trims" on page 20.

CONNECT MIDI GEAR

Connect your MIDI device's MIDI IN jack to the UltraLite-mk3's MIDI OUT jack (Connection A below). Conversely, connect the MIDI device's MIDI OUT jack to the UltraLite-mk3's MIDI IN jack (Connection B).
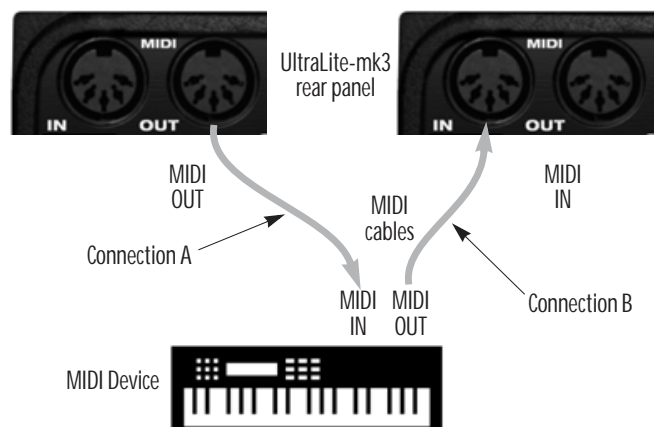

Figure 4-5: Connecting a MIDI device to the UltraLite-mk3.

INSTALLING THE ULTRALITE-MK3 HYBRID HARDWARE

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 4-5. Similarly, devices that never send data, such as a sound module, only need Connection A. Make both connections for any device that needs to both send and receive MIDI data.

Connecting additional gear with MIDI THRUs

If you need to connect several pieces of MIDI gear, run a MIDI cable from the MIDI THRU of a device already connected to the UltraLite-mk3 to the MIDI IN on the additional device as shown below in Figure 4-6. The two devices then share the UltraLite-mk3'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.
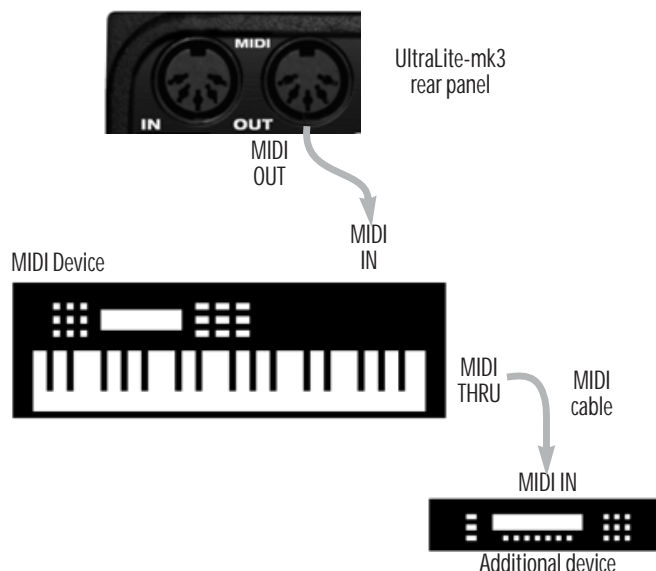

Figure 4-6: Connecting additional devices with MIDI THRU ports.

MIDI Thru when operating stand-alone

The UltraLite-mk3 has a MIDI Thru feature for stand-alone operation. See "MIDI Thru in standalone" on page 38.

CONNECTING AND SYNCING S/PDIF DEVICES

DAT decks and other devices with S/PDIF digital I/O will sync to the UltraLite-mk3 in via the S/PDIF connection itself. Just connect it to the UltraLite-mk3 via the S/PDIF connectors. When the device records a digital audio signal (from the UltraLite-mk3), it will simply synchronize to the clock provided by the digital audio input.

On the other hand, when you transfer audio from the S/PDIF device into the UltraLite-mk3, you'll have to slave the UltraLite-mk3 to its S/PDIF input.
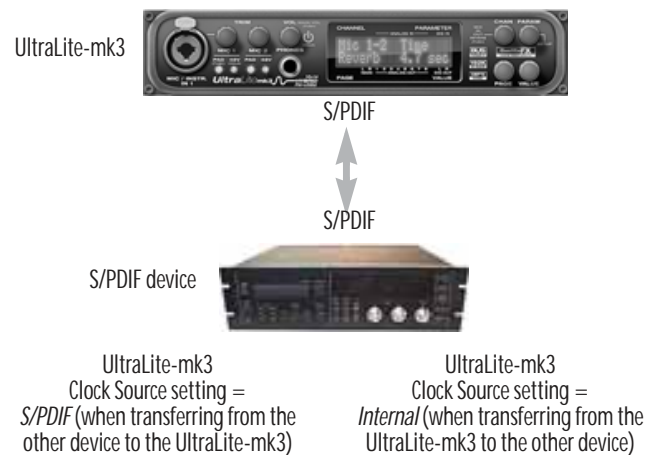

Figure 4-7: The setup for synchronizing a S/PDIF device with the UltraLite-mk3. Sync is achieved via the digital I/O connection itself. In this case, you have to choose S/PDIF as the UltraLite-mk3's clock source when recording from the other device.

POWER OPTIONS

The UltraLite-mk3 can draw power from two possible sources:

1. the computer via FireWire, or

2. a DC power supply.

If you use USB2 to connect to your computer, you must also use the DC power supply included with your UltraLite-mk3, as USB2 does not supply enough power by itself.

Bus power requirements

The UltraLite-mk3 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.

6-pin FireWire connectors

The UltraLite-mk3 can only draw power over the FireWire bus from a 6-pin to 6-pin cable, or a 6-pin to 9-pin (FireWire B) cable. It cannot draw power from a FireWire cable with a 4-pin connector, as shown below:
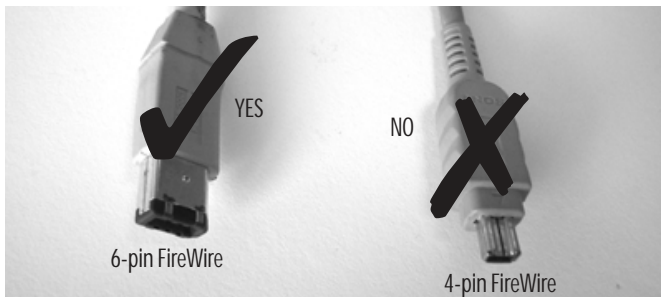

Figure 4-8: 4-pin FireWire connectors cannot be used for bus power.

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

The UltraLite-mk3 can be daisy-chained with other FireWire devices from a single FireWire connection to the computer. However, if the UltraLite-mk3 is operating under bus power, this is not recommended. If you need to daisy chain the UltraLite-mk3 with other devices on the same FireWire bus, power the UltraLite-mk3 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 UltraLite-mk3 to a PCI card and run the UltraLite-mk3 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 UltraLite-mk3 to a PC card FireWire adapter (inserted in the PC card slot in your laptop), it must provide a 6-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 UltraLite-mk3 operation:

Bus power from a desktop computer

Your desktop computer is running off of its usual AC power connection, and the UltraLite-mk3 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 UltraLite-mk3 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 UltraLite-mk3 is being powered by its FireWire connection to the computer. So the laptop battery is supplying power to both the laptop and the UltraLite-mk3. 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 UltraLite-mk3 to draw power from the computer, and AC power is available, you can power the UltraLite-mk3 from any standard 8-18 volt, 12 watt DC power supply with any polarity (tip positive or negative), and amperage as shown below.

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

Bus power versus DC power

When you connect a DC power supply, the UltraLite-mk3 could draw power from either FireWire or the DC power supply. So which source does it use? The answer is: the source that is currently supplying the greatest voltage. But in practice, it doesn't really matter because the only situation in which you need to be concerned about the UltraLite-mk3's power draw is when you are operating it with a battery-powered laptop. And in this situation, the only available power is from FireWire. In all other situations, the computer has its own power, so it doesn't matter if the UltraLite-mk3 is drawing power from FireWire.

Turning off the UltraLite-mk3

To turn on the UltraLite-mk3, push the VOL knob. To turn it off, push *and hold* the VOL knob. When the UltraLite-mk3 is turned off, it is really in a sort of "sleep" mode, where it still draws just enough power to detect the power switch (a digital encoder) when the UltraLite-mk3 is turned back on. But the amount of power that the UltraLite-mk3 draws when it is turned off is so small that it has very little practical impact. If you are running a laptop under battery power, and you are in a situation where you are not using the UltraLite-mk3 and you need every last bit of laptop battery power, unplug the UltraLite-mk3 entirely from the computer.

A TYPICAL ULTRALITE-MK3 HYBRID SETUP

Here is a typical UltraLite-mk3 studio setup. This rig can be operated without an external mixer. All mixing and processing can be done in the computer with audio software. During recording, you can use the UltraLite-mk3's CueMix™ FX no-latency monitoring to listen to what you are recording via the main outs, headphone outs, or any other output pair. You can control monitoring either from the front panel or from the included CueMix FX software.
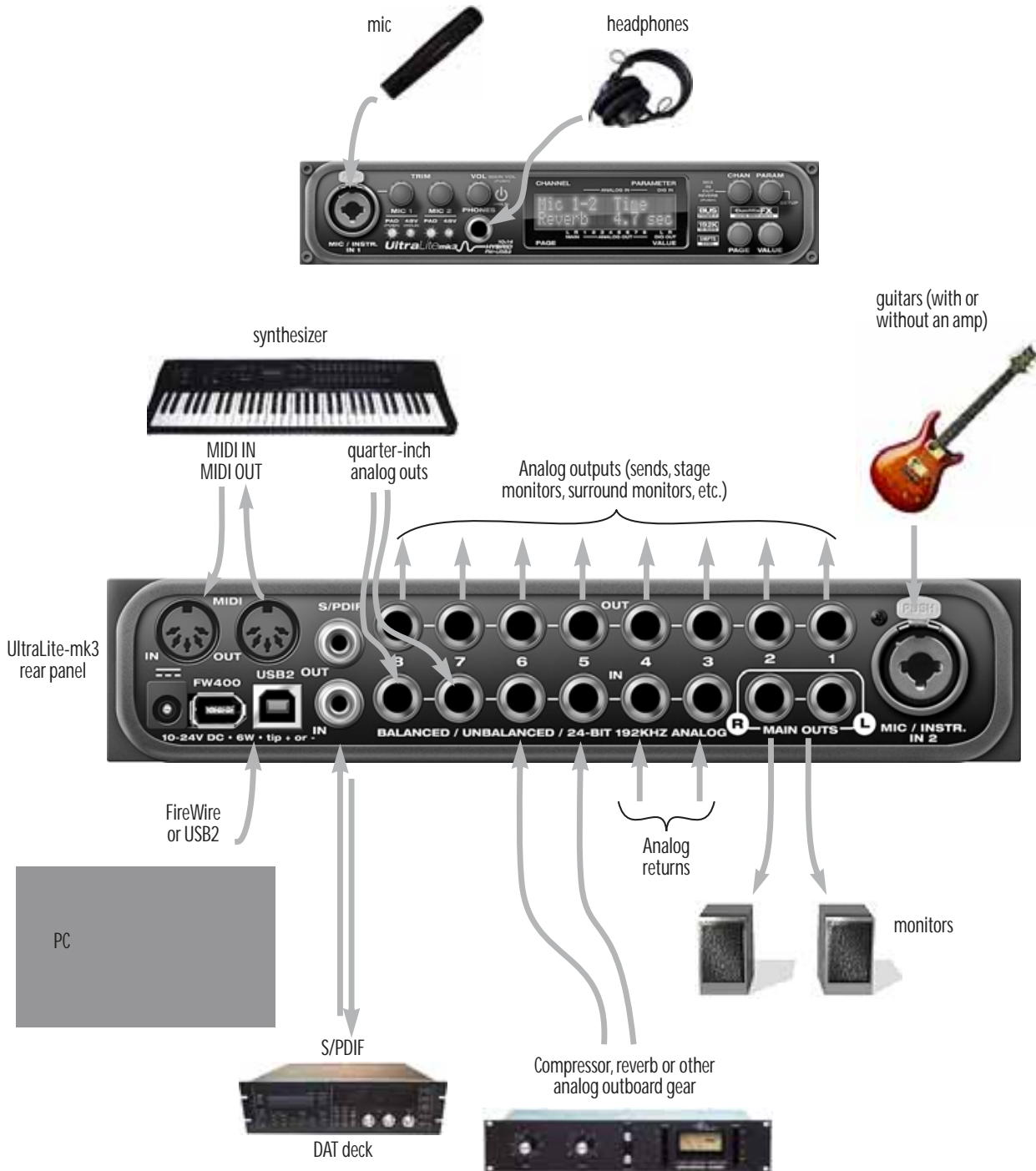

Figure 4-9: A typical UltraLite-mk3 studio setup.

CONNECTING MULTIPLE MOTU FIREWIRE INTERFACES

You can daisy-chain up to four MOTU FireWire interfaces on a single FireWire bus, with the restrictions described in the following sections. Most computers have only one built-in FireWire bus (even if it supplies multiple FireWire sockets).
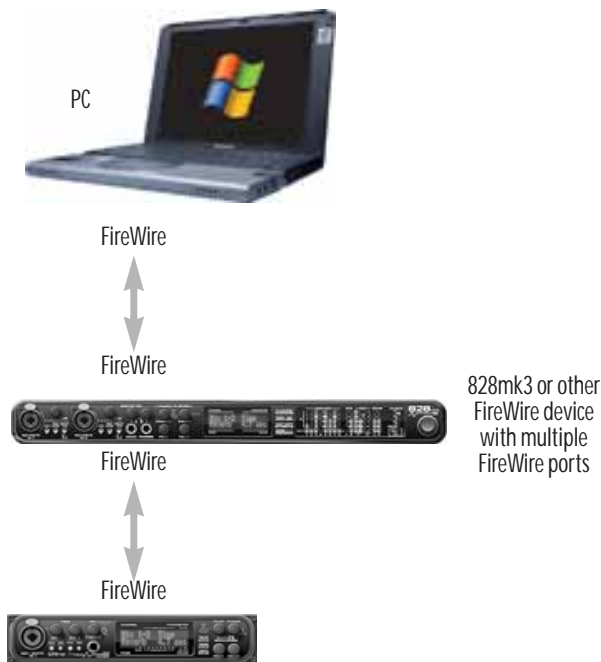

PC

FireWire

FireWire

828mk3 or other
FireWire device
with multiple
FireWire ports

FireWire

FireWire

Figure 4-10: Connecting multiple MOTU FireWire audio interfaces.

Multiple interfaces cannot be bus-powered
Do not run the UltraLite-mk3 under bus power when connecting it with other devices to the same FireWire bus. See "Power options" and "Bus power requirements" on page 23.

Multiple interfaces in MOTU Audio Console
MOTU Audio Console displays the settings for one interface at a time. To view the settings for an interface, click its tab as shown below in Figure 4-11.
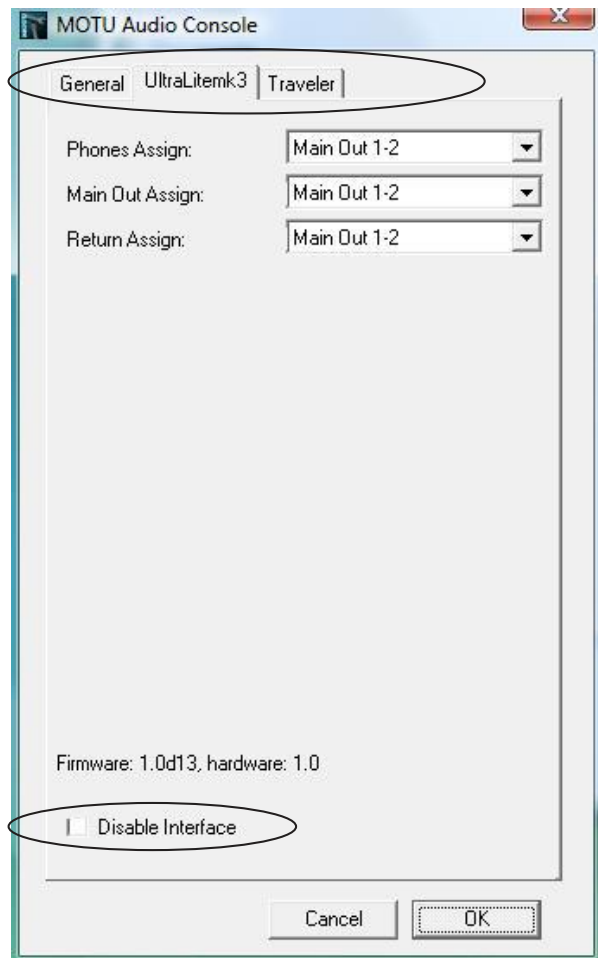

Figure 4-11: To view the settings for an interface, click its tab.

Synchronizing multiple interfaces
All connected MOTU FireWire interfaces get their clock from whatever you choose from the *Clock Source* menu in the *General* tab in MOTU Audio Console. When you connect multiple MOTU FireWire interfaces, all of their respective sync sources are displayed in the menu as shown below in Figure 4-12.
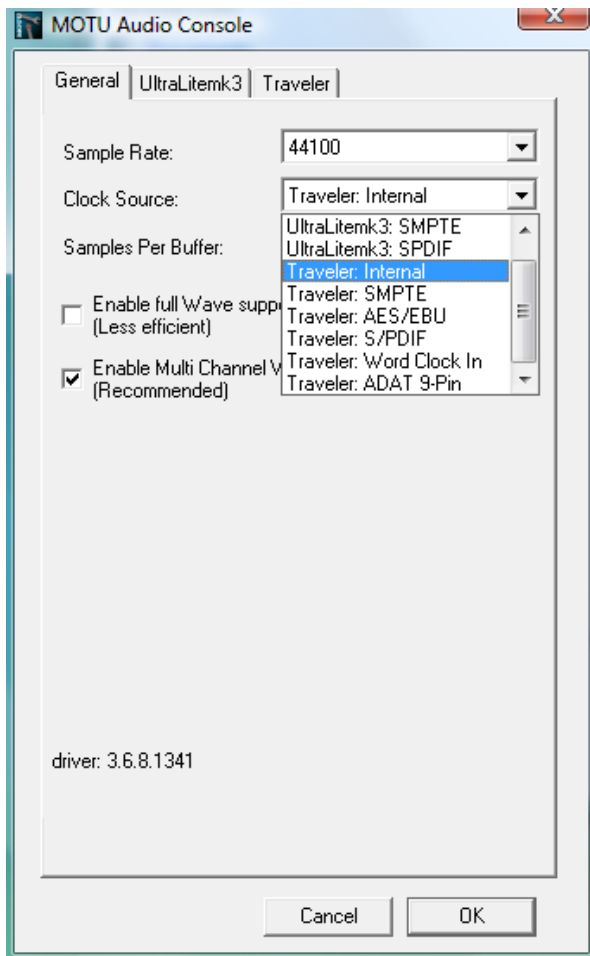
Figure 4-12: All MOTU FireWire audio interfaces get their clock from a single master sync source on any connected UltraLite-mk3 (or other MOTU FireWire interface). After you choose a source from this menu, the entire system, including all connected UltraLites, synchronizes to it.

Each FireWire interface in the system gets its clock from the computer (unless it is the master clock itself).

Connecting other MOTU FireWire interfaces

You can add an original MOTU 828 to the end of a FireWire daisy chain (because the 828 has only one FireWire port), or you can mix and match multiple UltraLites with other MOTU FireWire interfaces using a standard FireWire hub. You can also add 828mkIIs, 828mk3s, 896HDs, 896mk3s, Travelers and UltraLites, which have two FireWire ports convenient for daisy-chaining. Up to four interfaces can be combined on one FireWire bus.

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 (on interfaces that have them) to conserve FireWire bus bandwidth. At the 2x samples rates (88.2 or 96kHz) and 4x sample rates (176.4 and 192kHz), you can operate no more than two FireWire interfaces on a single FireWire bus.

Adding additional interfaces with a second FireWire bus

Third-party FireWire bus expansion products in the form of a cardbus ("PC card") adaptor, ExpressCard adaptor or PCI card allow you to add a second FireWire bus to your computer. In may be possible to add additional MOTU FireWire interfaces connected to such a third-party product, depending on the performance of the product and the performance of your host computer.

Managing the IDs of multiple interfaces

Multiple UltraLite-mk3 interfaces are identified by number (#1, #2, #3, etc.) Interfaces are ID'd (given a number) by the order in which they are first powered up after being connected. This information is stored in the MOTU Audio preferences file. Once ID'd, they retain the same number regardless of the order in which they are powered up. You can disable an interface at any time with the *Disable interface* option shown in Figure 4-11 on page 26. Doing so frees up the FireWire bandwidth required by the interface without turning it off. Switching off an interface accomplishes the same thing. To get MOTU Audio Console to forget about an interface entirely, you'll see a *Forget* button in MOTU Audio Console. Just click the *Forget* button and MOTU Audio Console will no longer consider the interface to be present but off line (turned off).

# CHAPTER 5   MOTU Audio Console

OVERVIEW

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

ACCESSING THE ULTRALITE-MK3 SETTINGS

There are several ways to access MOTU Audio Console settings:

■ From the Windows Start menu, choose *Programs>MOTU>MOTU Audio Console*

■ In Cubase or Nuendo, open the Device Setup window, click *VST Audio System* and choose *MOTU Audio ASIO* from the *Master ASIO Driver* menu as shown below. Then click the *MOTU Audio ASIO* item in the list and click the *Control Panel* button.
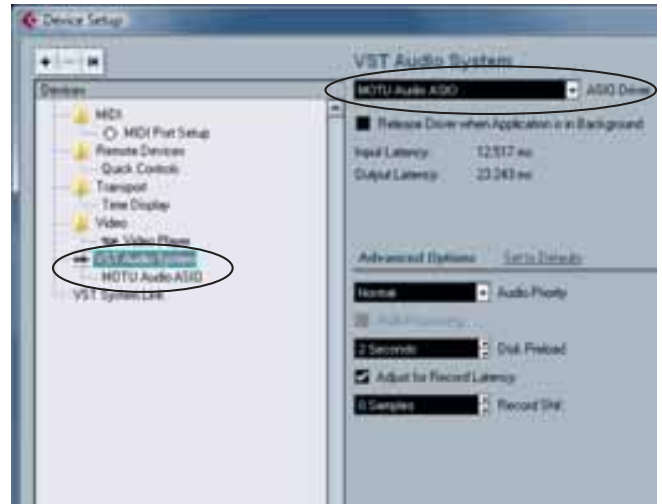
■ From within other ASIO-compatible programs, refer to their documentation.

■ From the front panel LCD (see chapter 6, "UltraLite-mk3 Hybrid Front Panel Operation" (page 35)).

General tab settings

The General tab provides settings that apply globally to all connected MOTU FireWire interfaces.

UltraLite-mk3 tab settings

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

'GENERAL' TAB SETTINGS

Sample Rate
Choose the desired *Sample Rate* for recording and playback. The UltraLite-mk3 can operate at 44.1 (the standard rate for compact disc audio), 48, 88.2, 96, 176.4 or 192KHz. If you have a S/PDIF device connected to the UltraLite-mk3, make sure that it matches the UltraLite-mk3's sample rate.

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

S/PDIF at 4x sample rates (176.4 or 192kHz)
At the 4x sample rates (176.4 or 192kHz), all analog I/O remains active, but S/PDIF digital I/O is disabled.

Clock Source
The *Clock Source* determines the digital audio clock that the UltraLite-mk3 will use as its time base. The following sections briefly discuss each clock source setting.

Internal
Use the *Internal* setting when you want the UltraLite-mk3 to operate under its own digital audio clock. For example, you may be in a situation where all you are doing is playing tracks off hard disk in your digital audio software on the computer. In a situation like this, you most often don't need to reference an external clock of any kind.

Another example is transferring a mix to DAT. You can operate the UltraLite-mk3 system on its internal clock, and then slave the DAT deck to the UltraLite-mk3 via the S/PDIF connection (usually DAT decks slave to their S/PDIF input when you choose the S/PDIF input as their record source).

If you would like help determining if this is the proper clock setting for your situation, see
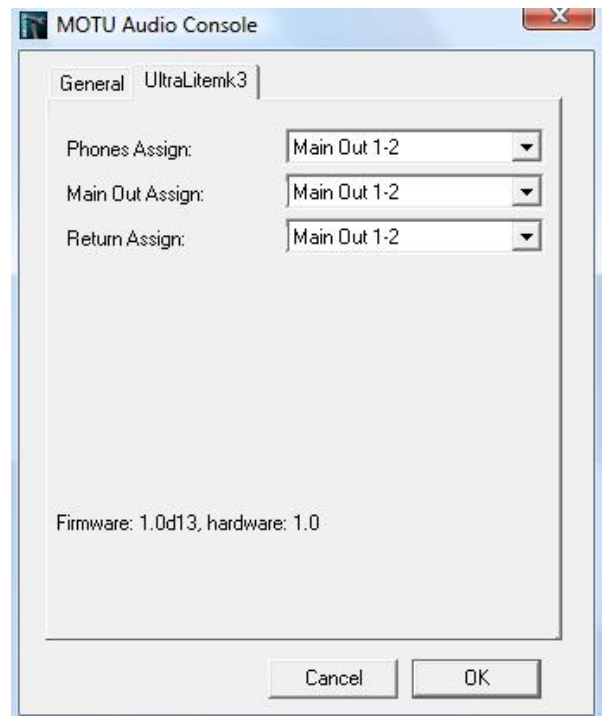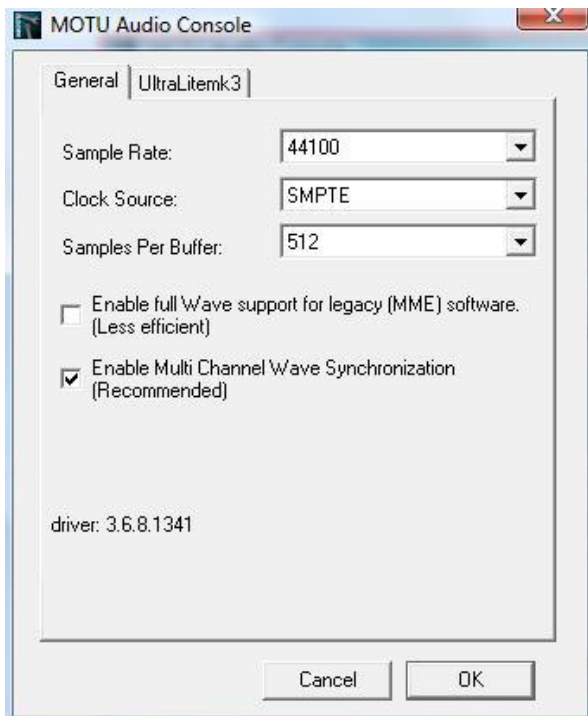

Figure 5-1: MOTU Audio Console gives you access to all of the settings in the UltraLite-mk3 hardware.

S/PDIF

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

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

For further details about this setting, see "Connecting and syncing S/PDIF devices" on page 22.

SMPTE

Choose this setting to resolve the UltraLite-mk3 directly to SMPTE time code (LTC) being received via the UltraLite-mk3's quarter-inch SMPTE input jack. For details, see "Syncing to SMPTE time code" on page 106and chapter 11, "MOTU SMPTE Console" (page 103).

Samples Per Buffer

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

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

☛ If you don't need to process an incoming live signal with software plug-ins, you can monitor the signal with no delay at all using CueMix FX, which routes the signal directly to your speakers via hardware. For details, see chapter 10, "CueMix FX" (page 61).

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

▪ The strain on your computer's CPU

▪ The delay you hear when routing a live signal through your host audio software plug-ins

▪ How responsive the transport controls are in your software

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

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

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

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

Monitoring live inputs without plug-in effects
As mentioned earlier, CueMix FX allows you to monitor dry, unprocessed live inputs with no delay at all. For complete details, see chapter 9, "Reducing Monitoring Latency" (page 55).

Wave support for legacy (MME) software
Windows only exposes the first two channels of a multi-channel WDM audio stream to applications which use the legacy (MME) multimedia interface. The *Enable full Wave support for legacy (MME) software (less efficient)* option forces the UltraLite-mk3 multimedia driver to expose all channels as stereo pairs, providing full MME support.

If your host audio software does *not* directly support WDM audio and instead only supports legacy MME drivers, use this option to access multiple UltraLite-mk3 input and output channels.

If your host audio software *does* directly support WDM audio, leave this option unchecked for optimal performance.

This option is only available when the multimedia driver has been installed, and it defaults to being not checked.

Enable Pedal
This setting applies to other MOTU FireWire audio interfaces, but it does not apply to the UltraLite-mk3.

'ULTRALITE-MK3' TAB SETTINGS

Phones Assign
The *Phones Assign* setting lets you choose what you will hear from the headphone jack. Choose *Main Out 1-2* if you'd like the headphone output to match the main outs. Choose *Phones* 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 eight on-board CueMix FX mix busses.

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 UltraLite-mk3 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 UltraLite-mk3 appears in your host software alongside all other UltraLite-mk3 inputs, wherever your host software lists them.

The UltraLite-mk3 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 UltraLite-mk3 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 UltraLite-mk3 hardware via CueMix FX (with or without CueMix effects processing on the live inputs).

Disable Interface

You can disable an interface at any time with the *Disable interface* option. Doing so frees up the FireWire bandwidth required by the interface without turning it off. Switching off an interface accomplishes the same thing. To get MOTU Audio Console to forget about an interface entirely, you'll see a *Forget* button in MOTU Audio Console. Just click the *Forget* button and MOTU Audio Console will no longer consider the interface to be present but off line (turned off).

UltraLite-mk3 Hybrid Front Panel Operation

OVERVIEW

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

PUSH-BUTTON ROTARY ENCODERS

All of the knobs on the UltraLite-mk3 front panel are push-button digital rotary encoders. In many cases, you can either push the knob, hold it in, or turn it to make a setting or toggle the LCD display (depending on the encoder and setting).

POWER SWITCH

Push the VOL knob to power on the UltraLite-mk3. Push in and hold the VOL knob to turn it off.

MIC/GUITAR INPUTS

The UltraLite-mk3 mic/guitar inputs are equipped with many features to handle a wide variety of recording situations.

For information about connections and settings, see "Mic/instrument inputs" on page 20 in the installation chapter.

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

- "The Inputs tab" on page 66

- "The channel settings section" on page 70

- "The Channel tab" on page 70

- "The EQ tab" on page 72

- "The Dynamics tab" on page 80

Figure 6-1: The UltraLite-mk3 front panel mic/guitar inputs and phone jack.

PHONES

From the factory, the PHONES jack (Figure 6-1) is a discrete output at 44.1/48 kHz, but it can mirror any other output pair (digital or analog) or serve as its own independent output.

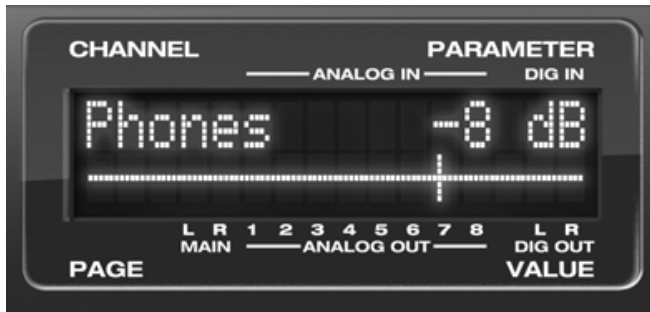When you turn the phone VOL knob, the LCD display provides visual feedback:

Figure 6-2: The LCD provides feedback as you adjust Phone volume.

To view the current setting without changing it, just push the knob (without turning it).

MAIN OUT VOLUME

Push the VOL knob twice to view and control the volume of the main outs on the rear panel.

Figure 6-3: The LCD provides feedback as you adjust Phone volume.

The MASTER volume adjustment can be programmed to control any combination of outputs. See "The Monitor Group" on page 84 for details.
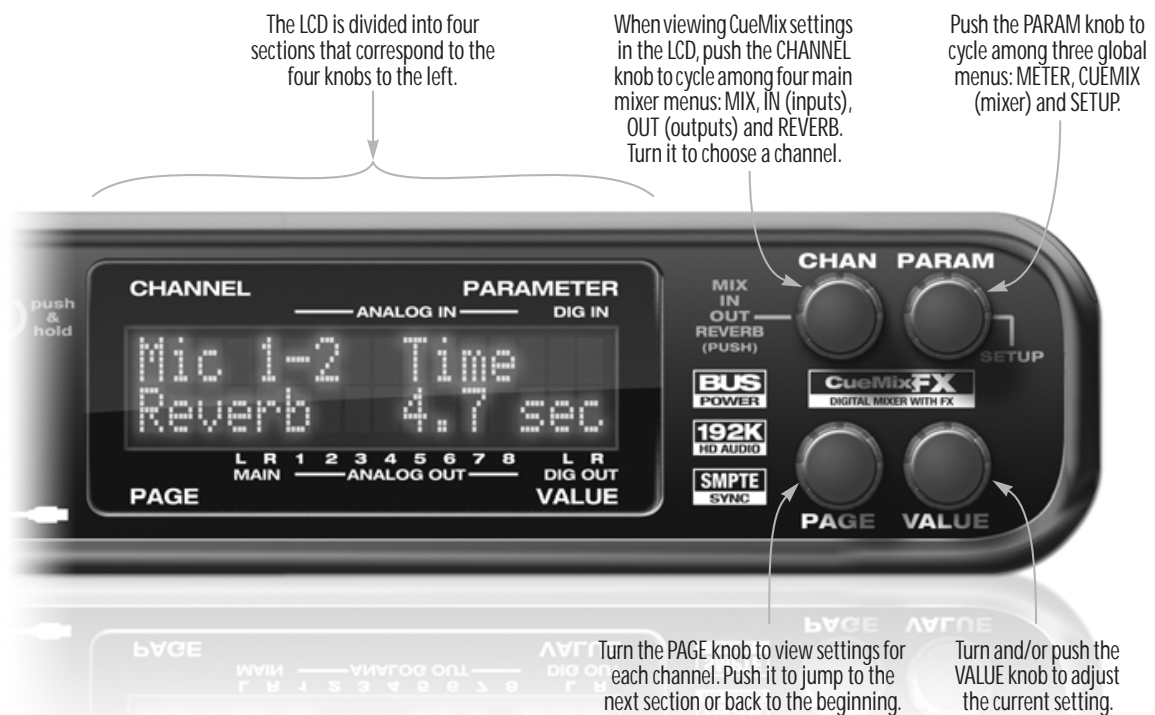
The LCD is divided into four sections that correspond to the four knobs to the left.

When viewing CueMix settings in the LCD, push the CHANNEL knob to cycle among four main mixer menus: MIX, IN (inputs), OUT (outputs) and REVERB. Turn it to choose a channel.

Push the PARAM knob to cycle among three global menus: METER, CUEMIX (mixer) and SETUP.

Turn the PAGE knob to view settings for each channel. Push it to jump to the next section or back to the beginning.

Turn and/or push the VALUE knob to adjust the current setting.

Figure 6-4: The UltraLite-mk3 front panel controls.

MULTI-FUNCTION LCD DISPLAY

The LCD provides access to the many UltraLite-mk3 settings, as well as visual feedback of the current parameter being modified.

Parameter "zooming"

For many settings, the LCD temporarily "zooms in" to display a long-throw meter and alpha-numeric display to give you precise, real-time feedback as you adjust the setting. For example, if you change the headphone volume, the LCD displays a level meter and gain reduction reading that updates as you turn the volume knob, as shown in Figure 6-2 on page 36. After a brief time-out, the display returns to its previous state before you turned the volume knob.

Four global display modes

Push the PARAM knob to cycle the LCD among four global display modes:

METER DISPLAY

To access the METER display, push the PARAM knob until you see *METER* displayed in the LCD. The METER display (Figure 6-5) provides comprehensive metering for each input and output. The labels above and below the LCD indicate the meters for each input and output pair, including the S/PDIF input and output (labeled DIG IN and DIG OUT, respectively).
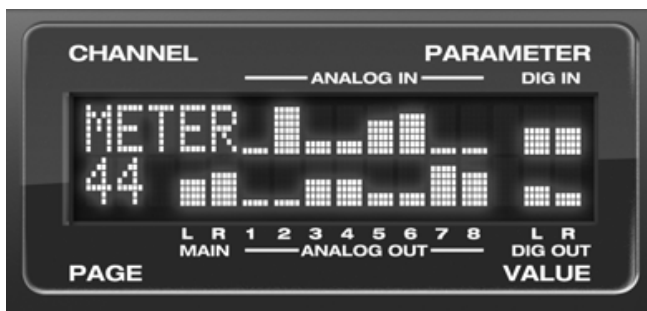

Figure 6-5: METER mode.

AUDIO MENU

To access the UltraLite-mk3 AUDIO menu, push the PARAM knob until you see *AUDIO* displayed in the LCD. This menu provides basic audio-related. Turn the PARAM knob to access each setting. Each setting in the AUDIO menu has a corresponding setting in the MOTU Audio Console software, as shown below:

Audio menu item	Where to find more information
Clock Source*	"Clock Source" on page 30
Sample rate*	"Sample Rate" on page 30
Phones Assign	"Phones Assign" on page 32
Main Out Assign	"Main Out Assign" on page 32
Return Assign	"Return Assign" on page 32

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

SETUP MENU

To access the UltraLite-mk3 SETUP menu, push the PARAM knob until you see *SETUP* displayed in the LCD. This menu provides basic features for managing the UltraLite-mk3 hardware. Turn the PARAM knob to access each setting. The first five settings in the SETUP menu have corresponding settings in the MOTU Audio Console software, as shown below:

LCD Contrast

Turn the VALUE knob to adjust the LCD contrast.

Save/Name Preset

An UltraLite-mk3 preset holds all current CueMix FX mix settings (everything in the CUEMIX menu). SETUP menu parameters are not included. Turn the VALUE knob to move from character to character in the preset name. Turn PAGE to change the currently flashing letter. Once you've named the preset, push VALUE to save it, turn it again to choose the desired preset slot you

wish to save it to (1-16), and the push again to confirm the save. To cancel the save operation at any time, turn the PARAM knob.

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

MIDI Thru in standalone
Turn VALUE to enable (Y) or disable (N) the UltraLite-mk3's *MIDI Thru* feature when the it is being operated stand-alone (not connected to a computer via FireWire). When MIDI Thru is enabled, the MIDI IN jack passes MIDI data directly to the MIDI OUT jack. This allows you, for example, to play a sound module connected to the MIDI OUT from a keyboard controller connected to the MIDI IN, without a computer connected.

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

AutoSave Status
All of the parameters in the UltraLite-mk3 are periodically saved automatically in its memory, so that if you power down the unit, it will power back up in the same state. AutoSave Status tells you the current state of this process: *Change Detected*, *Saving…* or *Saved*. If it says *Saved*, it is ready to be powered down. If it says *Change Detected* or *Saving…*, wait until it changes to *Saved*.

Factory Defaults
Push VALUE to restore the UltraLite-mk3 hardware to its factory default settings. Push VALUE again to confirm, or turn PARAM to cancel.

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

CueMix mixer basics
It is much easier to navigate the CUEMIX menu in the LCD if you have a general understanding of the CueMix FX mixer. We strongly recommend that you review chapter 10, "CueMix FX" (page 61) before learning the LCD, especially "CueMix FX basic operation" on page 63.

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

- IN (inputs)

- OUT (outputs)

- MIX (Mixes)

- REVERB (reverb processor)

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

The IN (inputs) menu

Push the CHANNEL button repeatedly until you see "I:" in the channel section of the LCD (Figure 6-6). This is similar to accessing the Inputs tab in CueMix FX console ("The Inputs tab" on page 66).
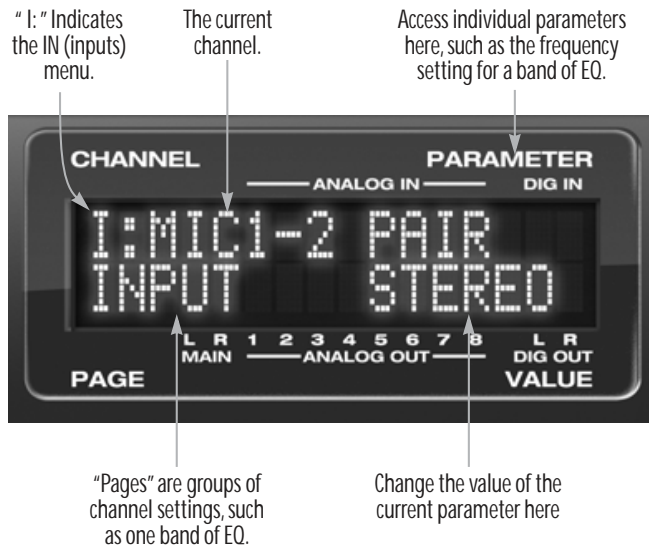

Figure 6-6: The IN (inputs) menu.

Choosing a channel

Once you see the Inputs menu (Figure 6-6) in the LCD, turn the CHANNEL knob to select the desired input that you wish to edit. This is roughly equivalent to specifying an input channel strip to work with in the Inputs tab in CueMix FX software (Figure 10-3 on page 66).

Choosing a setting to modify

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

Turn the PAGE knob to scroll through channel settings such as individual bands of EQ, the compressor, reverb sends, etc. Push the PAGE knob to jump to the next "section" of parameters or to jump back to the beginning of the list. This is roughly the equivalent of moving through the various channel controls in an individual input

channel strip in CueMix FX (Figure 10-3 on page 66), as well as the settings in the Channel tab (Figure 10-8 on page 70).

Turn the PARAMETER knob to scroll through individual parameters, such as the frequency for the current band of EQ. This is roughly equivalent to the parameters in the EQ tab (Figure 10-10 on page 72), Dynamics tab (Figure 10-26 on page 80) and Reverb tab (Figure 10-28 on page 83) in the CueMix FX software.

Adjusting the value of a parameter

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

Inactive items

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

Copying and pasting

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

Pad and phantom power

If a quarter-inch plug is inserted into the XLR/TRS combo input, the phantom power and pad settings are disabled. When disabled, their corresponding setting in the LCD menu will read *N/A* (Not Applicable). All current phantom power and pad settings for the mic input are reactivated and restored when the quarter-inch plug is removed from the jack.

Summary of Inputs menu settings

For a summary of Inputs menu settings, see "Inputs menu" on page 41.

The OUT (Outputs) menu

Push the CHANNEL button repeatedly until you see "O:" in the channel section of the LCD (Figure 6-7). This is similar to accessing the Outputs tab in CueMix FX console ("The Outputs tab" on page 68).
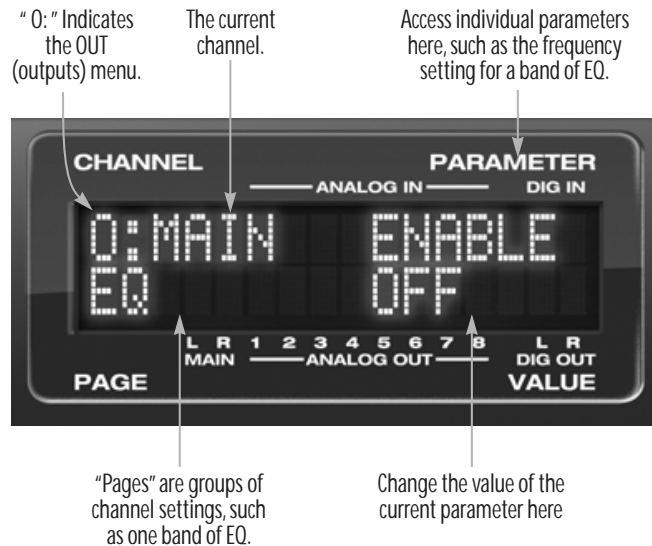

Figure 6-7: The OUT (outputs) menu.

Choosing a channel

Once you see the Outputs menu (Figure 6-7) in the LCD, turn the CHANNEL knob to select the desired output that you wish to edit. This is roughly equivalent to specifying an output channel strip to work with in the Outputs tab in CueMix FX software (Figure 10-6 on page 69).

Working with outputs in the LCD

Once you've selected an output channel, you can access the various settings for that channel using the PAGE knob and PARAMETER knobs. The knobs function the same as described for inputs in "Choosing a setting to modify" on page 39 and "Adjusting the value of a parameter" on page 39.

Summary of Output menu settings

For a summary of Inputs menu settings, see "Outputs menu" on page 42.

The MIX (Mixes) menu

Push the CHANNEL button repeatedly until you see "MIX 1" (or "MIX 2", etc.) in the channel section of the LCD (Figure 6-8). This is similar to accessing the Mixes tab in CueMix FX console ("The Mixes tab" on page 64).
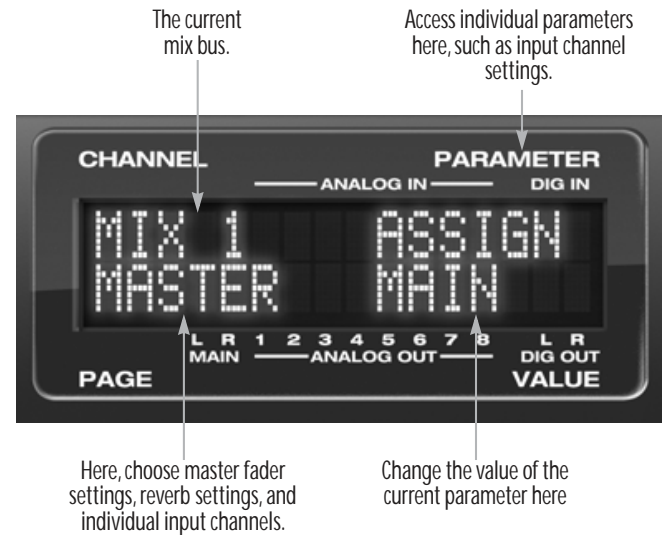

Figure 6-8: The MIX (Mixes) menu.

Choosing a mix bus

Once you see the MIX menu (Figure 6-8) in the LCD, turn the CHANNEL knob to select the desired mix that you wish to edit. This is roughly equivalent to choosing a mix in the Mixes tab in CueMix FX software (Figure 10-2 on page 64).

Working with mix busses in the LCD

Once you've selected a mix, you can access the various settings for that mix using the PAGE knob and PARAMETER knobs. The knobs function the same as described for inputs in "Choosing a setting to modify" on page 39 and "Adjusting the value of a parameter" on page 39. The PAGE knob scrolls through the following mix bus settings:

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

- REVERB — these are the reverb send and return controls for the bus master fader. Access them with the PARAMETER knob.

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

Summary of Mix menu settings

For a summary of Mix menu settings, see "Mixes menu" on page 42.

The REVERB settings

Push the CHANNEL button repeatedly until you see "REVERB" in the channel section of the LCD (Figure 6-9). This is similar to accessing the Reverb tab in CueMix FX ("The Reverb tab" on page 83).

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

Figure 6-9: The REVERB settings.

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

Summary of Reverb menu settings

See "Reverb menu" on page 42 for a summary.

INPUTS MENU

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

OUTPUTS MENU

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

MIXES MENU

CHANNEL	PAGE	PARAM
MIXES Mix 1 Mix 2 etc.	MASTER	ASSIGN MUTE FADER COPY PASTE RESET
	REVERB	SEND RETURN
	ANALOG 1 ANALOG 2 etc. (if mono)	MUTE SOLO (BAL/WID) - n/a PAN FADER
	ANALOG 3-4 ANALOG 5-6 etc. (if stereo)	MUTE SOLO BAL/WID BAL FADER
	SP (S/PDIF)	MUTE SOLO BAL/WID BAL FADER

REVERB MENU

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

STAND-ALONE OPERATION

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

CHAPTER 7 Cubase, Nuendo, Live and Other ASIO Software

OVERVIEW

The UltraLite-mk3 includes an ASIO driver that provides multi-channel I/O and sample-accurate synchronization with Steinberg's Cubase family of digital audio sequencers, including Cubase and Nuendo.

WHAT IS ASIO?

ASIO is an acronym for *Audio Streaming Input* and *Output*. The ASIO MOTU FireWire Audio driver allows the UltraLite-mk3 to provide multi-channel audio input and output for any audio application that supports ASIO drivers.

Attention: Other software users

The UltraLite-mk3 ASIO driver also provides multi-channel I/O with any ASIO-compatible audio software. Cubase is used for the examples in this chapter. However, the basic procedures are the same and can be easily applied to any ASIO-compatible software. Just follow the general descriptions at the beginning of each main section in this chapter. Consult your software documentation for details about each topic, if necessary.

If your audio software doesn't support ASIO

If your host audio software does not support ASIO, but instead supports WDM (or legacy Wave) audio drivers, refer to the next chapter.

PREPARATION

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

■ chapter 3, "IMPORTANT! Run the UltraLite-mk3 Software Installer First" (page 15)

■ chapter 4, "Installing the UltraLite-mk3 Hybrid Hardware" (page 17).

RUN MOTU AUDIO CONSOLE

Before you run Cubase, launch MOTU Audio Console to configure your UltraLite-mk3 hardware. MOTU Audio Console lets you configure your audio interface, and it lets you enable the desired inputs and outputs. Only enabled inputs and outputs will be available to Cubase, so this is an important step. For complete details regarding MOTU Audio Console, see chapter 5, "MOTU Audio Console" (page 29).
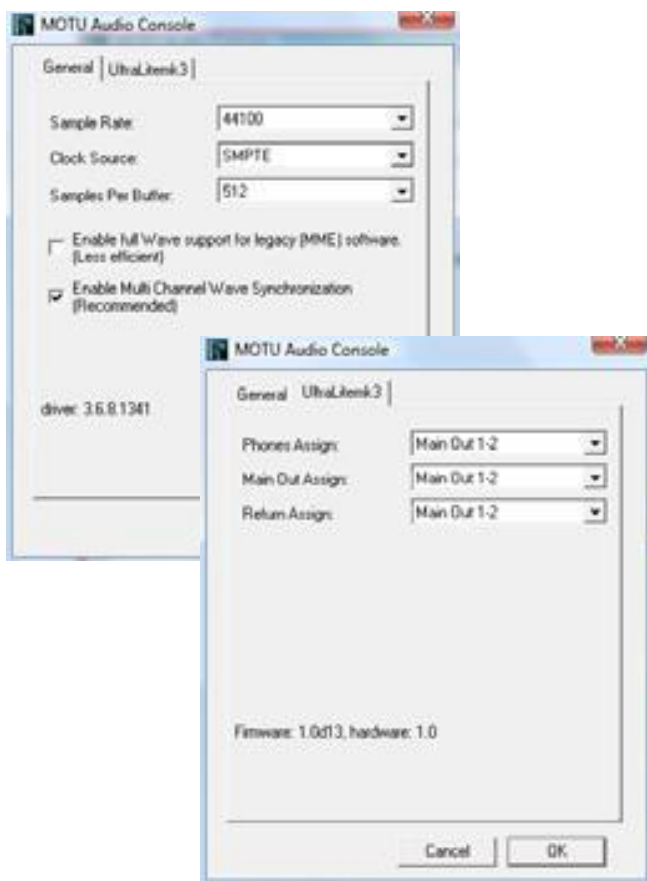
Figure 7-1: MOTU Audio Console gives you access to all of the settings in the UltraLite-mk3 hardware, including the clock source and sample rate.

For complete details about the UltraLite-mk3 settings, see chapter 5, "MOTU Audio Console" (page 29). The following sections provide a brief explanation of each UltraLite-mk3 setting for use with Cubase.

Sample rate
Choose the desired overall sample rate for the UltraLite-mk3 system and Cubase. Newly recorded audio in Cubase will have this sample rate.

Clock Source
This setting is very important because it determines which audio clock the UltraLite-mk3 will follow.

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

If you have a S/PDIF digital audio device connected to the UltraLite-mk3, see "Connecting and syncing S/PDIF devices" on page 22.

If you are slaving the UltraLite-mk3 and Cubase to SMPTE time code via the UltraLite-mk3 itself, choose *SMPTE* and follow the directions in "Resolving Cubase or Nuendo to SMPTE time code" on page 105.

☞ If you are using an ASIO host application other than Cubase or Nuendo, it must support the ASIO 2.0 sample-accurate positioning protocol in order to support the UltraLite-mk3's direct SMPTE sync (and sample-accurate sync) feature.

Samples Per Buffer
The *Samples Per Buffer* setting can be used to reduce the delay — or *monitoring latency* — that you hear when live audio is patched through your UltraLite-mk3 hardware and Cubase. For example, you might have MIDI instruments, samplers, microphones, and so on connected to the analog inputs of the UltraLite-mk3. If so, you will often be mixing their live input with audio material recorded in Cubase. See chapter 9, "Reducing Monitoring Latency" (page 55) for complete details.

Phones
This UltraLite-mk3 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 Cubase or Nuendo's audio output menus.

Main Out Assign

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

CHOOSING THE MOTU ASIO 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 ASIO 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 ASIO drivers that may be in your system. Choose the MOTU ASIO Driver from this menu.

Nuendo, Cubase and V-Stack

To activate the UltraLite-mk3 driver in Nuendo or Cubase SX, go to the Device Setup window, click *VST Audio System* and choose *MOTU Audio ASIO* from the *ASIO Driver* menu (Figure 7-2). Make the other settings in the dialog as need for your system and synchronization scenario.
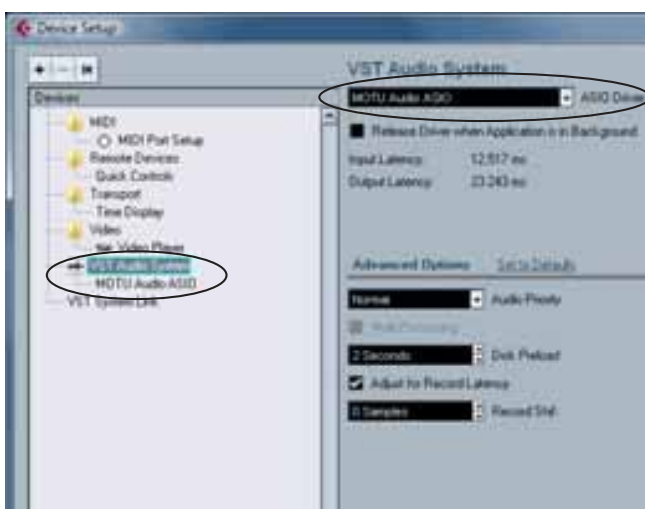
Figure 7-2: Activating the UltraLite-mk3 ASIO driver in Nuendo and Cubase.

Live

In Ableton Live, access the preferences window and click the *Audio* tab. Choose *ASIO* from the *Driver Type* menu. Choose the *MOTU Audio ASIO* from the *Audio Device* menu as shown below in Figure 7-3. To enable or disable UltraLite-mk3 input or output channels, click the *Input Config* or *Output Config* buttons. To access MOTU Audio Console, click the *Hardware Setup* button.

For information about the *Buffer Size* setting, see "Adjusting the audio I/O buffer" on page 57.
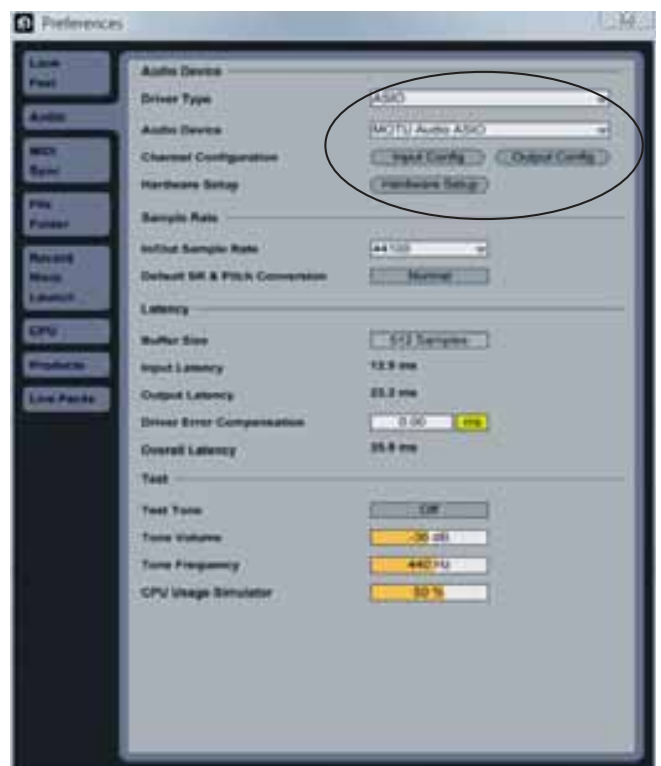

Figure 7-3: Enabling the UltraLite-mk3 in Live.

Sound Forge

To activate the MOTU Audio ASIO driver in Sound Forge, go to the Preferences window, click the *Audio* tab, and then choose *MOTU Audio* from the *Audio device type* menu as shown below.

Figure 7-4: Enabling the UltraLite-mk3 driver in Sound Forge.

DIRECT MONITORING

The *Direct Monitoring* option (Figure 7-5) allows you to monitor inputs directly in the UltraLite-mk3 hardware with no drain on your computer and near zero latency. When you enable this option, Cubase uses the UltraLite-mk3's CueMix FX monitoring features whenever you use Cubase's monitoring features. For further information, see "Controlling CueMix FX from within Cubase or Nuendo" on page 59.

OTHER SYSTEM DIALOG SETTINGS

Consult your Cubase or Nuendo documentation for details about the rest of the settings in this dialog.

VIEWING AND MANAGING ULTRALITE-MK3 INPUTS

Once you've chosen the MOTU ASIO Driver in Device Setup as explained earlier in "Choosing the MOTU ASIO driver" on page 45, click *MOTU Audio ASIO* in the left-hand list to see the UltraLite-mk3 inputs in the port list on the right (Figure 7-5). To activate an UltraLite-mk3 output, create a bus in the VST Connections window and assign the desired output to it.
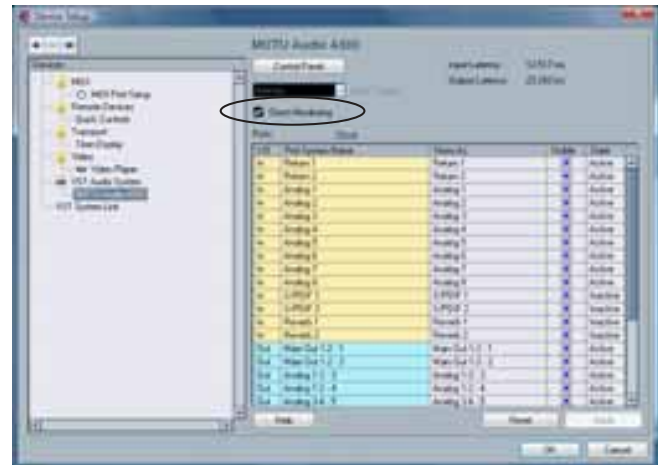
Figure 7-5: Creating UltraLite-mk3 inputs in Nuendo or Cubase.

Return Assign

In the VST Inputs list, you'll see an UltraLite-mk3 input pair called *Return 1-2*. This is a stereo feed from the UltraLite-mk3 that matches the output of one of its output pairs. Use the Return Assign menu in MOTU Audio Console to choose which output pair you would like to hear on this return. This can be used, for example, to record back a final stereo mix that includes effects processing from the UltraLite-mk3 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 UltraLite-mk3 output pair as the returns.

Reverb return

The UltraLite-mk3 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.

VIEWING AND MANAGING ULTRALITE-MK3 OUTPUTS

To view and manage UltraLite-mk3 outputs, scroll down below the inputs to see them (Figure 7-6).

Figure 7-6: Working with UltraLite-mk3 outputs in Nuendo or Cubase.

The "Phones 1-2" output

If you've chosen to treat the UltraLite-mk3 headphones as an independent output, you'll see *Phones 1-2* as a UltraLite-mk3 output destination. Audio tracks assigned to this output pair will be heard on the headphone jack only. For further explanation, see "Phones" on page 44.

CHANGING ULTRALITE-MK3 SETTINGS

To change the UltraLite-mk3 settings at any time, go to the Device Setup window in Nuendo or Cubase and click the Control Panel button, as shown in Figure 7-6 on page 47. Be sure to click the Reset button after making any changes.

PROCESSING LIVE INPUTS WITH HOST-BASED VST PLUG-INS

If you patch a live input (such as MIDI synthesizer) through a VST plug-in effect in Cubase, you might hear a slight delay. There are several ways to reduce this delay. For details, see chapter 9, "Reducing Monitoring Latency" (page 55).

WORKING WITH CUEMIX FX MIXING AND EFFECTS

The UltraLite-mk3 provides powerful external mixing, EQ, compression and reverb, which you can operate hand-in-hand with your host software's complete mixing environment. For example, the UltraLite-mk3 can serve as a monitor mixer routing channels to musicians, or it can serve as an integrated extension of your Cubase/Nuendo mixing environment. If you program an UltraLite-mk3 mixing and processing configuration that goes hand in hand with your Cubase/Nuendo project, be sure to use the file save features in CueMix FX to save the UltraLite-mk3 settings as a file in your Cubase/Nuendo project folder for instant recall of all settings. See chapter 10, "CueMix FX" (page 61) for complete details.

SYNCHRONIZATION

Together, Cubase (or Nuendo) and the UltraLite-mk3 have the ability to resolve directly to SMPTE time code, without an additional time code synchronizer. See "Syncing to SMPTE time code" on page 106.

MIDI I/O VIA THE ULTRALITE-MK3 MIDI PORTS

Once you've run the UltraLite-mk3 software installer as explained in "Installing the UltraLite-mk3 software" on page 15, the UltraLite-mk3 MIDI ports will appear as a MIDI input source and output destination in your host software's MIDI input and output menus.

24-BIT OPERATION

Your UltraLite-mk3 hardware fully supports
Cubase and Nuendo's 24-bit recording capabilities.
Simply enable 24-bit operation as instructed in
your Cubase or Nuendo manual. The
UltraLite-mk3 always supplies a 24-bit data
stream, and when you enable 24-bit operation in
Cubase or Nuendo, it simply uses all 24-bits
supplied by the UltraLite-mk3 hardware.

SONAR and other WDM Software

OVERVIEW

The UltraLite-mk3 WDM driver provides standard multi-channel input and output for WDM-compatible audio software running under Windows XP.

WDM AND WAVE DRIVER COMPATIBILITY

WDM is an acronym for *Windows Driver Model.* The MOTU WDM Driver allows the UltraLite-mk3 to provide multi-channel audio input and output for any audio application that supports WDM audio drivers.

Attention: Cakewalk users

Cakewalk SONAR is used for the WDM driver setup examples in this chapter.

Attention: other software users

SONAR is used for the WDM driver setup examples in this chapter. However, the basic procedures are the same and can be easily applied to any WDM-compatible software. Consult your software documentation for details, if necessary.

INSTALLING THE MOTU AUDIO WDM DRIVER

The MOTU Audio Installer CD installs the MOTU WDM driver into Windows for you. That's pretty much all the preparation you need. See chapter 3, "IMPORTANT! Run the UltraLite-mk3 Software Installer First" (page 15).

ENABLING WAVE (MME) COMPATIBILITY

If your host audio software does not support Windows WDM drivers, enable Wave driver compatibility in the MOTU WDM Driver so that all of the UltraLite-mk3's inputs and outputs show up in your software. For details, see "Wave support for legacy (MME) software" on page 32.

☛ If your host audio software *does* support WDM audio drivers, *don't* enable Wave driver compatibility.

WaveRT SUPPORT

The UltraLite-mk3 Windows driver also supports WaveRT, a low-latency audio driver standard developed by Microsoft. WaveRT is supported by Sonar 8 and later, under Windows 7 and Vista. To enable WaveRT, check the *Use WaveRT for Windows Audio* option in MOTU Audio Setup.

Figure 8-1: The WaveRT option can be enabled in the MOTU Audio Setup console.

If you uncheck this box, WaveRT support is disabled, and legacy WDM driver support is provided instead.

MAKING SETTINGS IN MOTU AUDIO CONSOLE

Before you run your audio software, launch MOTU Audio Console to configure your UltraLite-mk3 hardware. MOTU Audio Console lets you choose the audio clock source, sample rate, buffer size and other hardware settings. For details, see chapter 5, "MOTU Audio Console" (page 29). The following sections provide a brief explanation of each UltraLite-mk3 setting for use with SONAR.
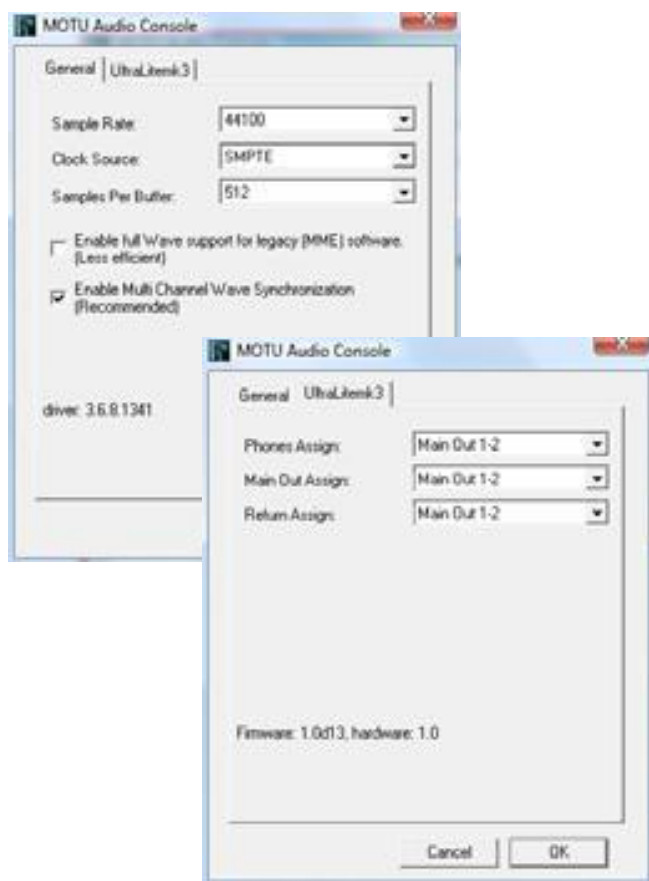
Figure 8-2: MOTU Audio Console gives you access to all of the settings in the UltraLite-mk3 hardware, including the clock source, sample rate and headphone output assignment.

Sample rate

Choose the desired overall sample rate for the UltraLite-mk3 system and SONAR. Newly recorded audio in SONAR will have this sample rate.

Clock Source

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

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

If you have a S/PDIF digital audio device connected to the UltraLite-mk3, see "Connecting and syncing S/PDIF devices" on page 22.

Samples Per Buffer

The *Samples Per Buffer* setting can be used to reduce the delay — or *monitoring latency* — that you hear when live audio is patched through your UltraLite-mk3 hardware and SONAR. For example, you might have MIDI instruments, samplers, microphones, and so on connected to the analog inputs of the UltraLite-mk3. If so, you will often be mixing their live input with audio material recorded in SONAR. See chapter 9, "Reducing Monitoring Latency" (page 55) for complete details.

Phones

This UltraLite-mk3 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*, 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 SONAR's audio output menus.

Main Out Assign

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

ENABLING THE MOTU AUDIO WDM 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 WDM driver. Check the audio system or audio hardware configuration window in your software.

Enabling the MOTU Audio WDM driver in SONAR

To activate the MOTU Audio WDM driver in SONAR:

1 Choose *Audio* from the Options menu.

2 Click the *Drivers* tab.

3 Check the UltraLite-mk3 inputs and outputs that you wish to use and uncheck the ones you don't as shown in Figure 8-3.
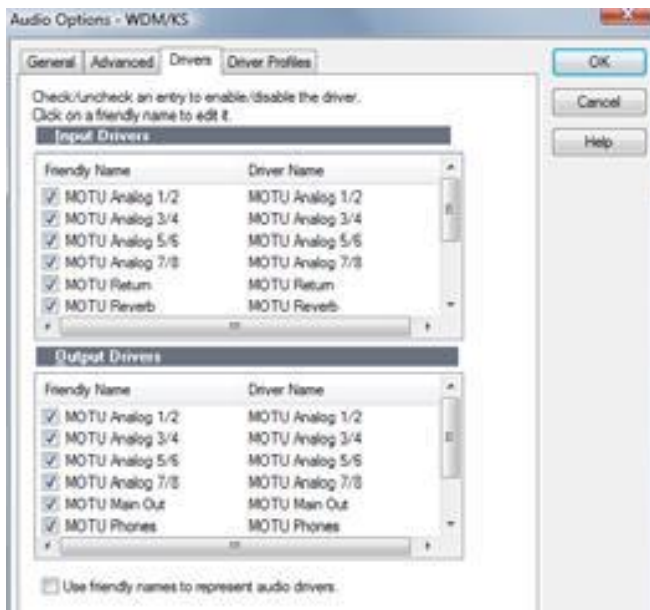
Figure 8-3: Activating the MOTU Audio WDM driver in SONAR by highlighting UltraLite-mk3 inputs and outputs.

Enabling the MOTU ASIO driver in SONAR

As an alternative to the UltraLite-mk3's WDM driver, you can use the UltraLite-mk3's ASIO driver. Note, however, that you can only use one audio interface at a time under ASIO. Go to the Advanced tab and choose *ASIO* from the *Driver Mode* menu:

Figure 8-4: Enabling the ASIO driver instead of the WDM driver.

Other audio settings in SONAR

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

1 In the Audio Options window, click the *General* tab.

2 Choose any UltraLite-mk3 input and output for recording and playback timing as shown below in Figure 8-5. It doesn't matter which ones.
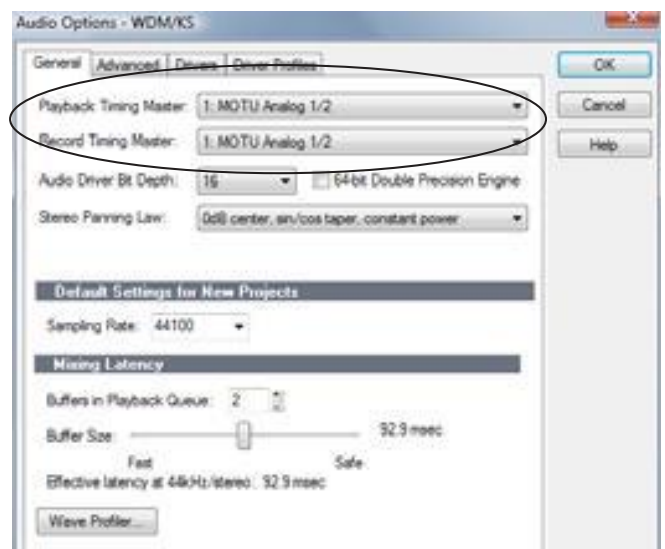

Figure 8-5: Make sure you have chosen an UltraLite-mk3 input and output for the playback and recording timing master settings.

3 In the General tab, click the Wave Profiler button and run the Wave Profiler.

This process chooses the optimum settings for the UltraLite-mk3 hardware.

WORKING WITH ULTRALITE-MK3 INPUTS AND OUTPUTS

Once you've enabled the UltraLite-mk3 inputs and output, UltraLite-mk3 audio inputs and outputs will appear in your host software's input/output menus, and you can set them up and use them as any standard audio input and output.

Return Assign

In SONAR, you'll see an UltraLite-mk3 input pair called *Return 1-2*. This is a stereo feed from the UltraLite-mk3 that matches the output of one of its output pairs. Use the Return Assign menu in MOTU Audio Console to choose which output pair you would like to hear on this return. This can be used, for example, to record back a final stereo mix that includes effects processing from the UltraLite-mk3 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 UltraLite-mk3 output pair as the returns.

Reverb return

The UltraLite-mk3 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.

The "Phones 1-2" output

If you've chosen to treat the UltraLite-mk3 headphones as an independent output, you'll see *Phones 1-2* as an UltraLite-mk3 output destination. Audio tracks assigned to this output pair will be heard on the headphone jack only. For further explanation, see "Phones Assign" on page 32.

CHANGING ULTRALITE-MK3 SETTINGS

You can change the UltraLite-mk3 settings at any time by accessing MOTU Audio Console.

REDUCING DELAY WHEN MONITORING LIVE INPUTS

If you have live audio inputs connected to the UltraLite-mk3, such as MIDI synthesizers, samplers, microphones or other live instruments, you might hear a slight delay when their audio is being monitored through your UltraLite-mk3 hardware and your host audio program. There are several ways to reduce — and eliminate — this audible monitoring delay. For details, see chapter 9, "Reducing Monitoring Latency" (page 55).

PROCESSING LIVE INPUTS WITH HOST-BASED VST PLUG-INS

If you patch a live input (such as MIDI synthesizer) through a plug-in effect in SONAR, you might hear a slight delay. There are several ways to reduce this delay. For details, see chapter 9, "Reducing Monitoring Latency" (page 55).

WORKING WITH CUEMIX FX MIXING AND EFFECTS

The UltraLite-mk3 provides powerful external mixing, EQ, compression and reverb, which you can operate hand-in-hand with your host software's complete mixing environment. For example, the UltraLite-mk3 can serve as a monitor mixer routing channels to musicians, or it can serve as an integrated extension of your SONAR mixing environment. If you program an UltraLite-mk3 mixing and processing configuration that goes hand in hand with your SONAR project, be sure to use the file save features in CueMix FX to save the UltraLite-mk3 settings as a file in your SONAR project folder for instant recall of all settings. See chapter 10, "CueMix FX" (page 61) for complete details.

MIDI I/O VIA THE ULTRALITE-MK3 MIDI PORTS

Once you've run the UltraLite-mk3 software installer as explained in "Installing the UltraLite-mk3 software" on page 15, the UltraLite-mk3 MIDI ports will appear as a MIDI

input source and output destination in your WDM-compatible software's MIDI input and output menus.

24-BIT OPERATION

Your UltraLite-mk3 hardware fully supports 24-bit recording in any audio software that supports it. Simply enable 24-bit operation as instructed by the software. The UltraLite-mk3 system always supplies the software with a 24-bit data stream, and when you enable 24-bit operation, it simply uses all 24-bits supplied by the UltraLite-mk3 hardware.

CHAPTER 9   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 mic 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 UltraLite-mk3 input, passes through the UltraLite-mk3 hardware into the computer, through your host audio software, and then back out to an UltraLite-mk3 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 UltraLite-mk3's CueMix FX digital mixer to patch the input directly to your monitor outs via the UltraLite-mk3 audio hardware. The UltraLite-mk3 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 58.

If you *do* need to process a live input with plug-ins, or if you are playing virtual instruments live through your UltraLite-mk3 audio hardware, you can significantly reduce latency — and even make it completely inaudible, regardless of what host audio application software you use. This chapter explains how.

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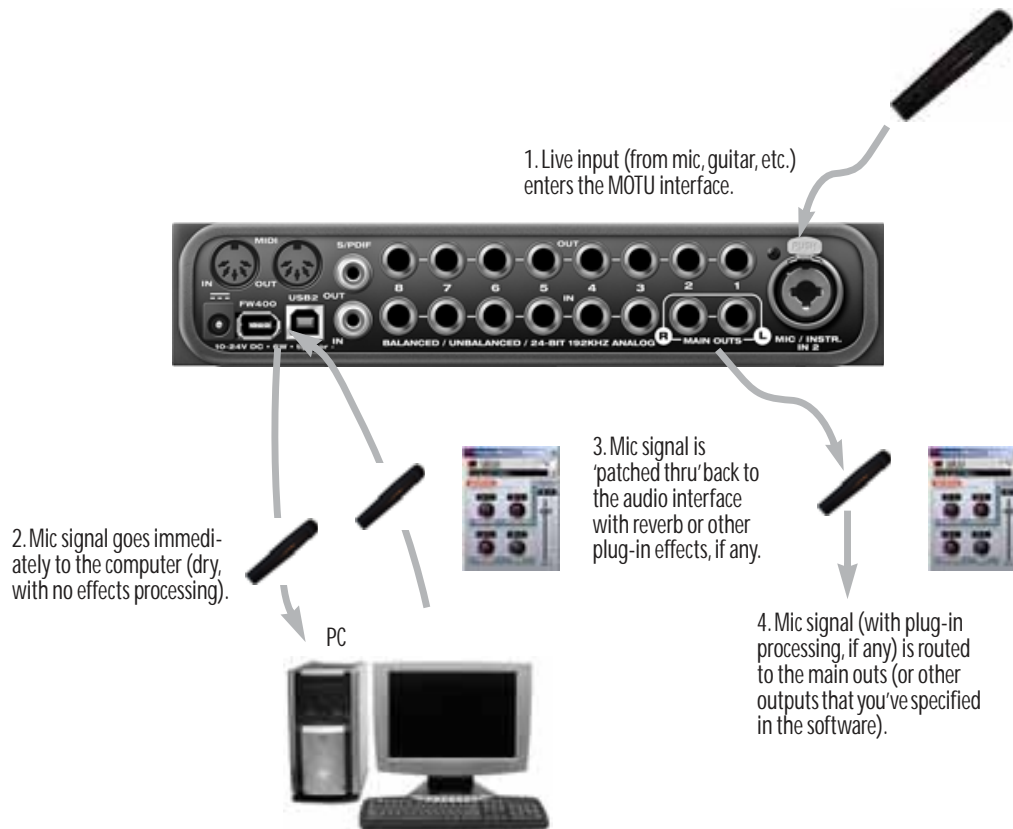
MONITORING LIVE INPUT

There are two ways to monitor live audio input with an UltraLite-mk3: 1) through the computer or 2) via the Ultralite-mk3's CueMix FX hardware mixer. Figure 9-1 below shows method 1, which allows you to add effects processing such as reverb and guitar amp effects 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 9-2 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 UltraLite-mk3-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 9-1 and Figure 9-2) 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 UltraLite-mk3 Classic Reverb.

1. Live input (from mic, guitar, etc.) enters the MOTU interface.

2. Mic signal goes immediately to the computer (dry, with no effects processing).

PC

3. Mic signal is 'patched thru' back to the audio interface with reverb or other plug-in effects, if any.

4. Mic signal (with plug-in processing, if any) is routed to the main outs (or other outputs that you've specified in the software).

Figure 9-1: There are two ways to monitor live audio inputs with an UltraLite-mk3: 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 UltraLite-mk3, buffers are used for the process of transferring audio data in and out of the computer. The size of the buffers determines how much delay you hear when monitoring live inputs through your audio software: larger buffers produce more delay; smaller buffers produce less.

Buffer size adjustment is made in MOTU Audio Console, as shown in Figure 9-3 via the *Samples Per Buffer* setting.
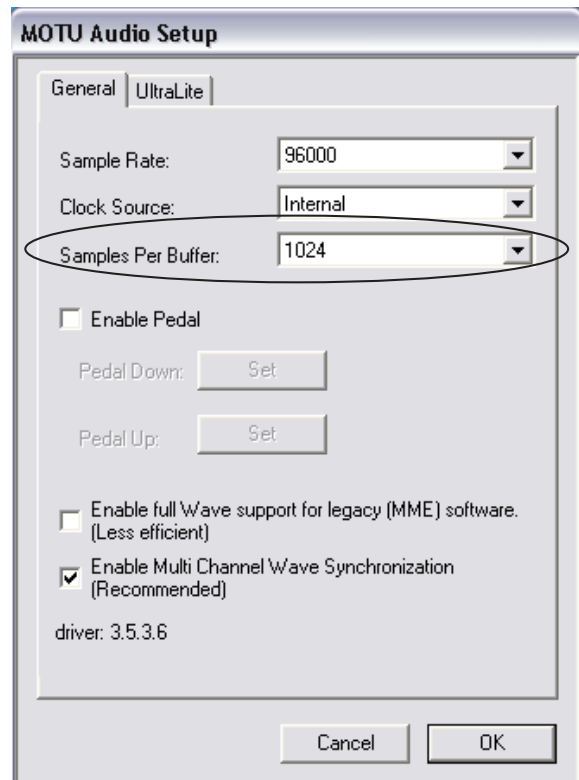
Figure 9-3: Lowering the 'Samples Per Buffer' setting in MOTU Audio Console reduces patch thru latency. But doing so increases the processing load on your computer, so keep an eye on the Performance Monitor in your host audio software.

1. Live input (from mic, guitar, etc.) enters the MOTU interface.

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.

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 UltraLite-mk3 effects processing, such as EQ, compression or Classic Reverb.

Figure 9-2: 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 UltraLite-mk3 effects, such as the reverb, EQ and/or compression.

Lower latency versus higher CPU overhead

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

■ Patch thru latency

■ The load on your computer's CPU

■ Possible distortion at the smallest settings

■ How responsive the transport controls are in your audio software

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

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

Transport responsiveness

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

Effects processing and automated mixing

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

CUEMIX FX HARDWARE MONITORING

The UltraLite-mk3 has a more direct method of patching audio through the system. This method is called *CueMix FX*. When enabled, CueMix activates hardware patch-thru in the UltraLite-mk3 itself. CueMix FX has two important benefits:

■ First, it completely eliminates the patch thru delay (reducing it to a small number of samples — about the same amount as one of today's digital mixers).

■ Secondly, CueMix FX imposes no strain on the computer.

The trade-off, however, is that CueMix FX bypasses your host audio software. Instead, live audio inputs are patched directly through to outputs in the UltraLite-mk3 itself and are mixed with disk tracks playing back from your audio software. This means that you cannot apply 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 DSP is the way to go.

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

TWO METHODS FOR CONTROLLING CUEMIX FX

There are two ways to control CueMix FX:

■ With CueMix FX

■ From within your host audio software (if it supports direct hardware monitoring)

You can even use both methods simultaneously.

Using CueMix FX

If your host audio software does not support direct hardware monitoring, you run CueMix FX 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 UltraLite-mk3 monitor mixes, or any other desired routing configurations. These routings are independent of your host audio software. For complete details, see chapter 10, "CueMix FX" (page 61).

Controlling CueMix FX from your audio software

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

CueMix FX routings that are made via host applications are made "under the hood", which means that you won't see them in CueMix FX. However, CueMix FX connections made inside your host audio software dovetail with any other

mixes you've set up in CueMix FX. For example, if your host application routes audio to an output pair that is already being used in CueMix FX for an entirely separate mix bus, both audio streams will simply be merged to the output.

Controlling CueMix FX from within Cubase or Nuendo

To turn on CueMix in Cubase SX or Nuendo, enable the Direct Monitoring check box in the Device Setup window (Figure 7-2 on page 45).

Other ASIO 2.0-compatible host software

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

Using CueMix FX with WDM- or Wave-compatible software

Run CueMix FX (chapter 10, "CueMix FX" (page 61)) and use it to route live inputs directly to outputs, control their volume and panning, etc. If necessary, you can save your CueMix Console setup with your project file.

CHAPTER 10 CueMix FX

OVERVIEW

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

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

For information about programming the UltraLite-mk3's on-board mixing and effects using the front panel LCD, see chapter 10, "CueMix FX" (page 61).

A 16-BUS MIXER WITH EQ, COMPRESSION AND REVERB

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

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

Effects include:

- Classic Reverb with tail lengths up to 60 seconds

- 7-band parametric EQ modeled after British analog console EQs

- A standard compressor with conventional threshold/ratio/attack/release/gain controls

- The Leveler™, an accurate model of the legendary LA-2A optical compressor, which provides vintage, musical automatic gain control

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

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

ADVANTAGES OVER HOST-BASED MIXING AND PROCESSING

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

■ CueMix has no buffer latency. Thanks to the UltraLite-mk3'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 UltraLite-mk3 to operate as a portable, stand-alone mixer with effects.

CUEMIX FX INSTALLATION

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

Channel focus · DSP resources meter · Mix bus menu · Solo light · Tabs for inputs, mix busses and outputs · UltraLite-mk3 inputs · Mix bus master fader · Channel settings · Monitoring/talkback section

Channel scroll bar

Mic inputs · Inputs split into mono channels · Inputs grouped as stereo pairs · Tabs for channel strip settings, including EQ and dynamics, as well as global settings such as the meter bridge and reverb processor. · Monitor group metering

Figure 10-1: CueMix FX is a virtual mixer that gives you control over the UltraLite-mk3's on-board mixing features.

CUEMIX FX BASIC OPERATION

Here is a brief overview of the CueMix FX mixer.

Eight stereo mix busses

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

Many inputs to one output pair

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

Viewing one mix bus at a time

CueMix FX displays one mix bus at a time in the *Mixes* tab (Figure 10-2 on page 64). To select which mix you are viewing, choose it from the mix bus menu (Figure 10-2). The mix name appears above the mix bus master fader (Figure 10-2), 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 10-3 on page 66) gives you access to settings for individual UltraLite-mk3 inputs (or input pairs), such as phase, trim, EQ and dynamics processing. Each input also includes a send to the UltraLite-mk3'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 10-6 on page 69) gives you access to settings for each UltraLite-mk3 output pair, including EQ, dynamics processing and send/return controls for feeding and returning the output signal to/from the UltraLite-mk3'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 10-1) to view channel-specific parameters in the Channel Settings section of the CueMix FX window (Figure 10-1). 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 UltraLite-mk3 has a global reverb module (Figure 10-28 on page 83). Once it has been activated, you can feed signals to the reverb processor from various points in the UltraLite-mk3 mix matrix via input sends, bus sends and output sends. Stereo output from the reverb processor can then be fed back to mix busses or output pairs using reverb returns.

Other features

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

Widening the CueMix FX window

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

THE MIXES TAB

Click the *Mixes* tab (Figure 10-2) to gain access to the UltraLite-mk3'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 10-2). The menu shows all mixes by name, followed by the UltraLite-mk3 output pair to which each bus master fader is currently assigned, if any.

Naming a mix

Click the mix name at the top of the mix bus master fader (Figure 10-2) to edit the name.

Assigning a mix bus output

Choose the desired output pair for the mix bus from the bus output menu (Figure 10-2). The bus output menu displays all current available (enabled) UltraLite-mk3 output pairs. If a bus is already assigned to an output pair, the bus name appears next to the output pair name to indicate that the output pair is already taken by a bus. Only one bus can be assigned to any given output pair. If you choose an output already assigned to another bus, that bus output will become disabled.

Bus fader

The bus fader (Figure 10-2) controls the overall level of the mix (its volume on its stereo output). Use the individual input faders to the left to control individual input levels.
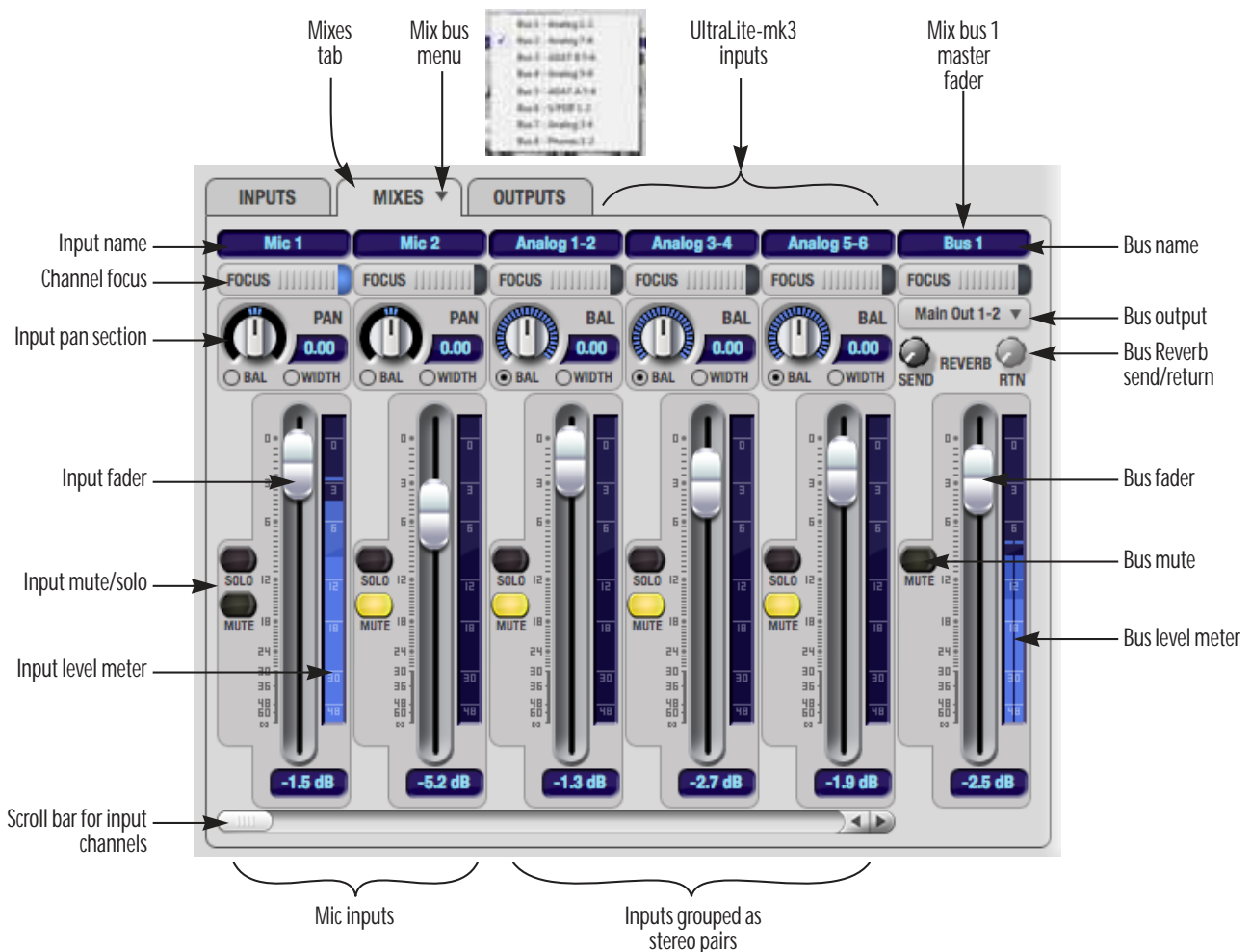
Figure 10-2: The Mixes tab.

Bus mute

The bus mute button (Figure 10-2) disables (silences) the mix.

Bus level meter

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

Bus reverb send/return

The bus reverb send (Figure 10-2) feeds the output of the mix bus, pre-fader, to the UltraLite-mk3'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 10-2) feeds the output of the UltraLite-mk3'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 83.

Input section

The horizontally scrolling area in the Mix tab to the left of the master fader (Figure 10-2) displays channel strips for all currently enabled UltraLite-mk3 inputs.

Input channel focus

Click the channel focus button (Figure 10-3) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 10-7 on page 70). Clicking the mix bus master fader focus button brings the assigned output into focus, if there is one.

Input pan section

The input pan knob (Figure 10-2) 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 10-2) 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 10-1) will illuminate.

THE INPUTS TAB

The UltraLite-mk3 provides many features for managing analog and digital input signals. Some of these features, such as the UltraLite-mk3'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 10-3) to access and control all of these input channel settings for each UltraLite-mk3 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 UltraLite-mk3'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, phase, stereo versus M/S decoding, width, L/R swap, EQ, dynamics and reverb send.

Input channel focus

Click the channel focus button (Figure 10-3) to view and edit parameters in the channel settings section of the window (Figure 10-7 on page 70).

Mono/stereo pairing

Click the *Mono* button (Figure 10-3) 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).

Figure 10-3: The Inputs tab.

Invert phase

The *Phase* button (Figure 10-3) inverts the phase of the input signal. For stereo pairs, you can invert the phase for the left and right channels independently.

Input trim

All UltraLite-mk3 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 two mic/guitar inputs and the six quarter-inch analog inputs on the back panel. Here is a summary of input trim ranges for each type of UltraLite-mk3 input:

Input	Trim cut	Trim boost	Trim Range
Mic/Guitar	0 dB	24 dB	24 dB
TRS analog inputs	-96 dB	+22 dB	118 dB
S/PDIF (RCA)	0 dB	+12 dB	12 dB

Once you adjust the trim levels, you can save them as a file on disk for future instant recall. See "Saving and loading hardware presets" on page 87 and "Configurations menu" on page 100.

Input EQ and dynamics

The UltraLite-mk3 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 10-3) 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 10-1). 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 70.

The EQ/Dynamics graph

The EQ/Dynamics graph for each input channel strip (Figure 10-3) 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 10-10 on page 72).

EQ/Dynamics selectors

The EQ/Dynamics selector buttons along the right-hand edge of the EQ/Dynamics section (Figure 10-3) allow you to choose what you are viewing and editing in the EQ/Dynamics section.
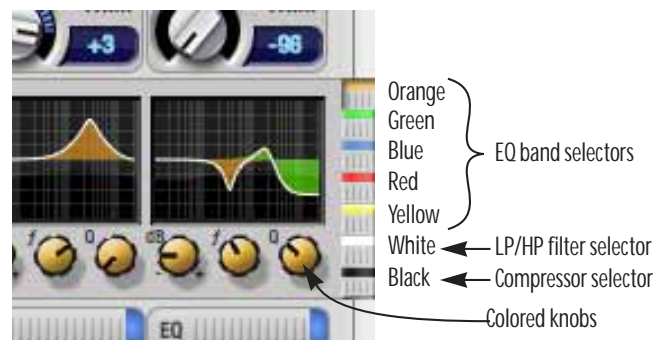
Figure 10-4: The EQ/Dynamics selectors.

Click the selector (Figure 10-4) for the desired EQ band, low-pass (LP) filter, high pass (HP) filter or compressor to view it across all channels.

Compressor graph   Compressor meter   Compressor selector
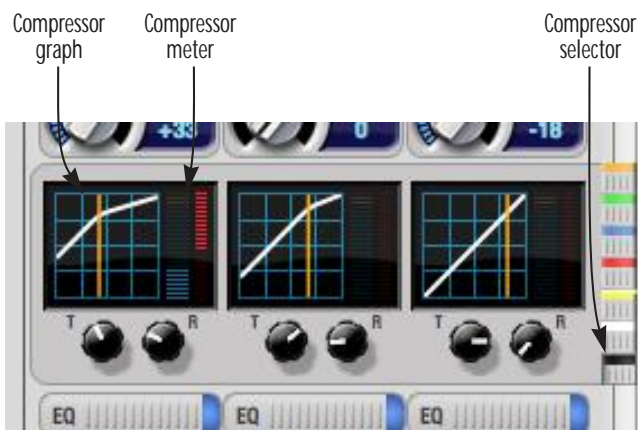
Figure 10-5: The Compressor controls.

☞ Shortcut: hold down the option 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 10-10), as explained in "Enabling EQ" on page 72.

EQ/Dynamics enable/disable buttons
Click the EQ or Dynamics button at the bottom of the input channel (Figure 10-3) 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 10-3) feeds the input signal to the UltraLite-mk3'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 10-6) lets you apply EQ, dynamics and reverb to any output pair, just before the signal leaves the UltraLite-mk3. This is processing that occurs at the very end of the signal flow, after everything else (host based effects, UltraLite-mk3 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 channel focus
Click the channel focus button (Figure 10-6) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 10-7 on page 70).

Output EQ and Dynamics
The EQ/Dynamics section in the Outputs tab (Figure 10-6) works identically to the EQ/Dynamics section for the Inputs tab (Figure 10-3). See "Input EQ and dynamics" on page 67.

Output reverb send/return
The output reverb *send* (Figure 10-6) feeds the signal for the output to the UltraLite-mk3'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 83.

The output reverb *return* (Figure 10-6) feeds the output of the UltraLite-mk3'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 10-6) to toggle whether the output pair is included in the Talkback or Listenback group. See "Talkback and listenback" on page 85.

Monitor group assign

Click the Monitor buttons (Figure 10-6) to toggle whether the output pair is included in the Monitor group. See "The Monitor Group" on page 84.
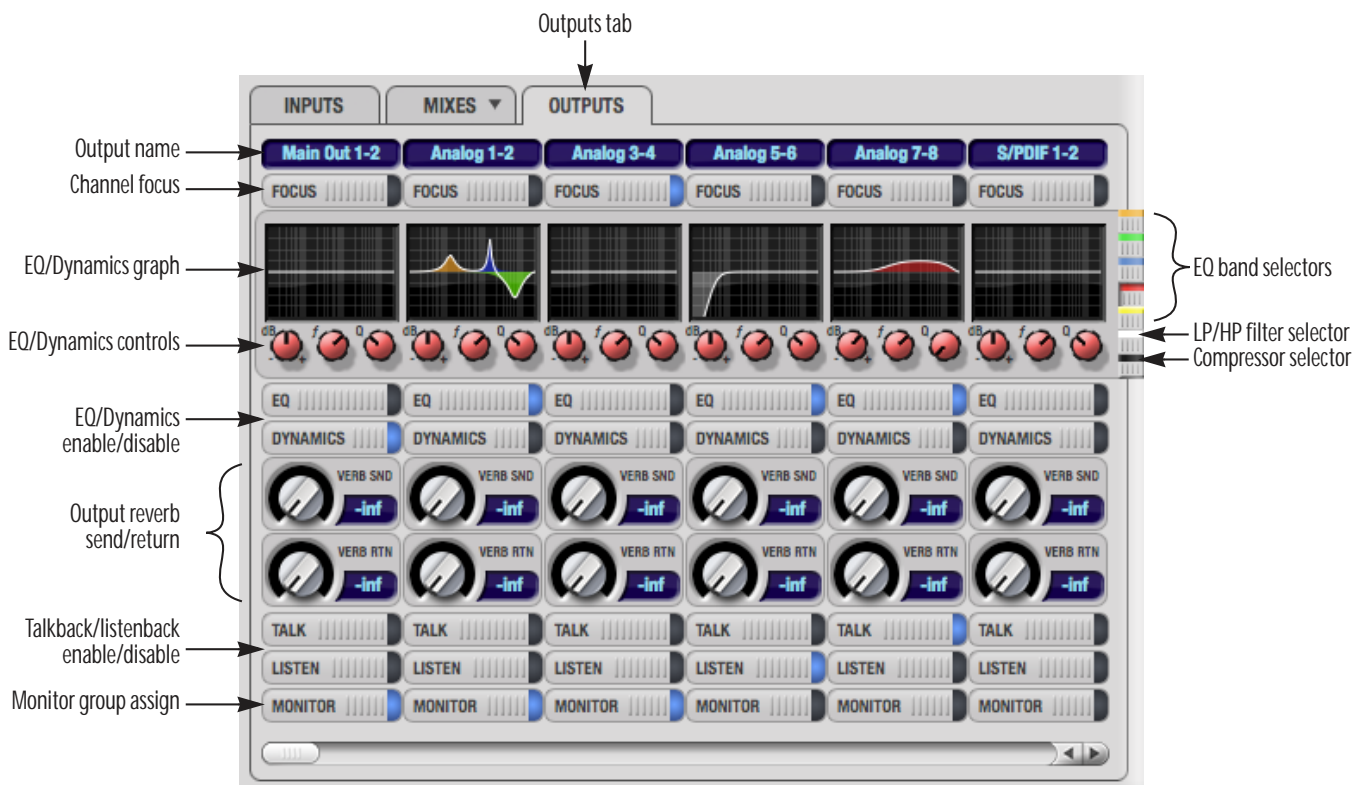
Figure 10-6: The Outputs tab.

C U E M I X   F X

THE CHANNEL SETTINGS SECTION

The channel settings section in the CueMix FX window (Figure 10-1) 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.
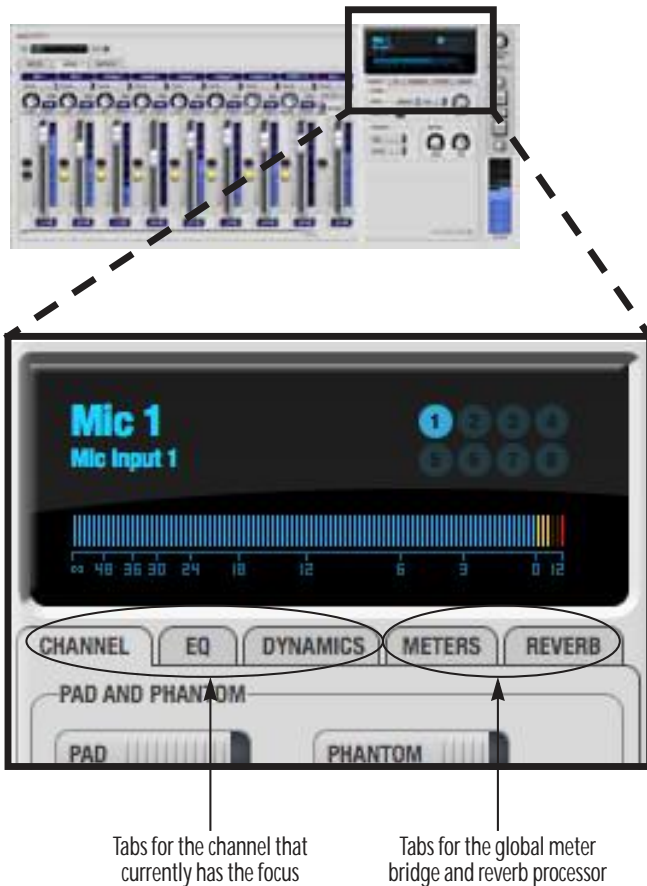
Figure 10-7: The Channel Settings section.

Tabs for the channel that currently has the focus

Tabs for the global meter bridge and reverb processor

The Channel tab

The Channel tab (Figure 10-8) displays settings for input channels. Click any focus button in the Inputs tab to view the Channel tab settings for the channel.

Figure 10-8: The Channel tab.

Signal flow

Settings in the Channel tab occur just before the EQ, dynamics and reverb sends in the Input tab channel strip (Figure 10-3 on page 66). 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 both mic inputs). These are the same 20 dB pad and 48V phantom power settings that you can also control from the UltraLite-mk3 front panel TRIM encoders. There are separate settings for each mic input.

Stereo settings

Inputs that have been grouped as stereo pairs in the Inputs tab (Figure 10-3) provide two stereo modes (Figure 10-8): *Normal* and *M/S*. M/S mode provides decoding for a mid-side microphone configuration.

The *Width* knob (Figure 10-8) provides control over the stereo imaging, going from a full stereo image to mono (both channels panned equally). See "Width" on page 65.

The *Swap L/R* button (Figure 10-8) lets you switch the left and right channels.

Talkback section

Click the *Talkback* or *Listenback* button (Figure 10-8) 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 85.

Reverb section

The *Send* in the reverb section (Figure 10-8) is the same control as the reverb send in the Input tab channel strip (Figure 10-3). See "Reverb send" on page 68. 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 10-8) to pan the mono signal for the stereo reverb processor.

Input meter and bus activity LEDs

When the Channel tab is active (Figure 10-8), the display above the tab provides a horizontal level meter and eight *bus activity LEDs* (Figure 10-9).
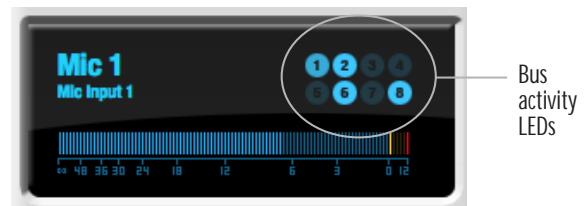
Figure 10-9: Input meter and bus activity LEDs.

The *input level meter* (Figure 10-9) is the same as the input meters in the Meters tab (Figure 10-27 on page 82) 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 UltraLite-mk3. 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 10-9) 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 10-10) 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 UltraLite-mk3 *Vintage EQ section* (Figure 10-10) 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 UltraLite-mk3 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 10-3 and Figure 10-6). 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 10-10), allowing you to enable as few or as many bands as needed for each individual channel.
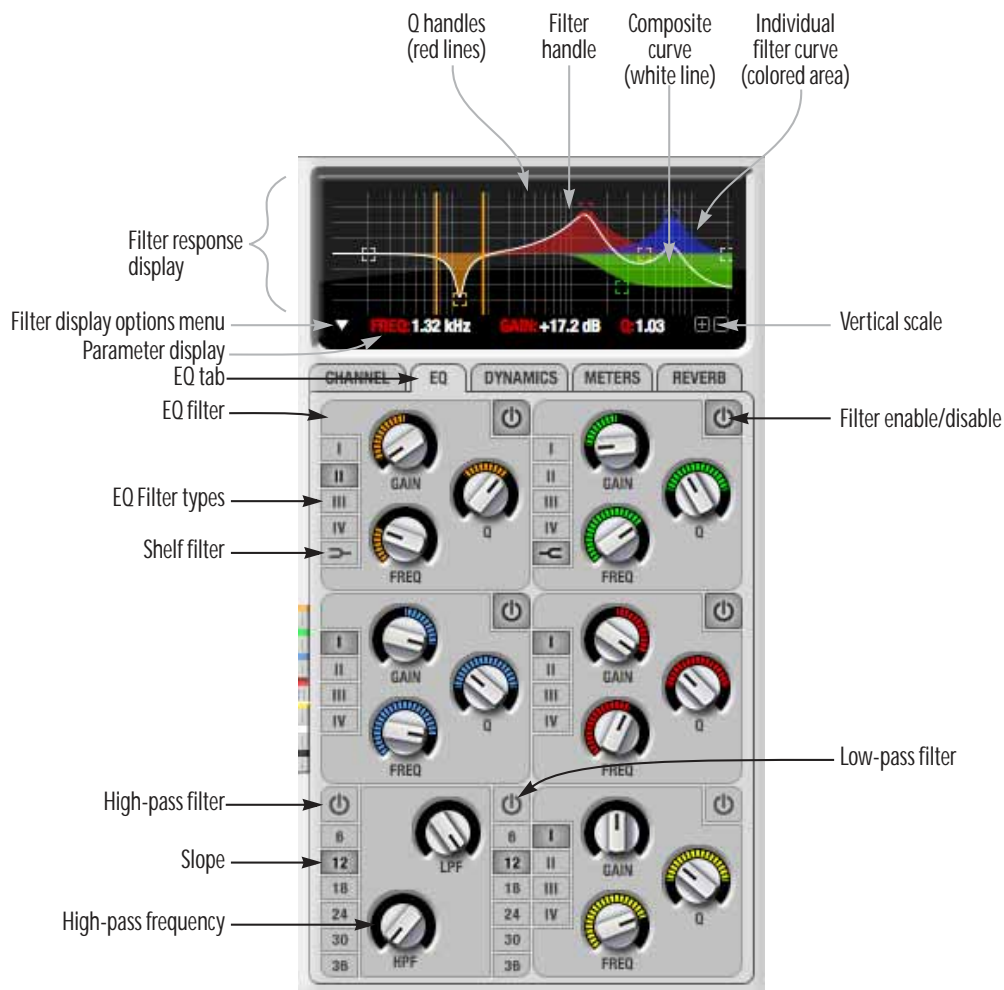
Figure 10-10: The EQ tab.

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Vintage EQ Quick reference

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

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

Parameter display: Shows the precise numbers of the parameter you are adjusting (or hovering over with the arrow cursor). The labels (*frequency, gain*, etc.) match the color of the filter being displayed. When a filter handle is not selected and when the cursor is not hovering over the display, the parameter display shows the name of the current channel being edited (the channel that currently has the focus), as shown below:
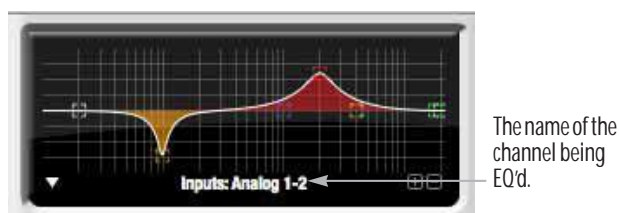
Figure 10-11: When a filter handle is not selected and when the cursor is not hovering over the display, the parameter display shows the name of the current channel being edited (the channel that currently has the focus).

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

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

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

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

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

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

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

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

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

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

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

Each filter can be independently turned on or off with the enable/disable button (Figure 10-10). 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 10-10).

Adjusting filters in the display

Each filter has a handle, displayed as shown below in Figure 10-12 (in the filter's color), for adjusting its boost/cut and/or frequency:
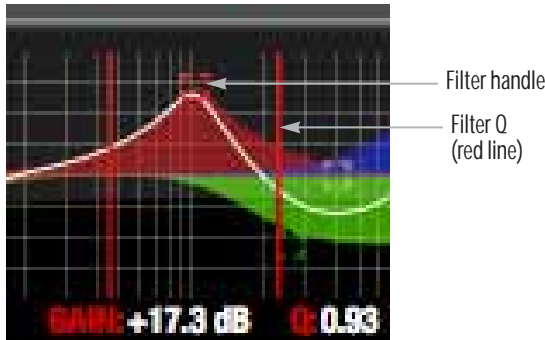
Figure 10-12: Drag the filter handle to adjust its frequency and/or boost/cut. Drag the Filter Q handles to adjust the Q.

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

Filter display options

The *Filter display options* menu (Figure 10-10) provides several options for the EQ filter display:

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

FFT display

Choose *Show FFT* from the *Filter display options* menu (Figure 10-10) to superimpose a real-time Fast Fourier Transform (FFT) frequency measurement curve over the EQ filter display, as demonstrated in Figure 10-13:
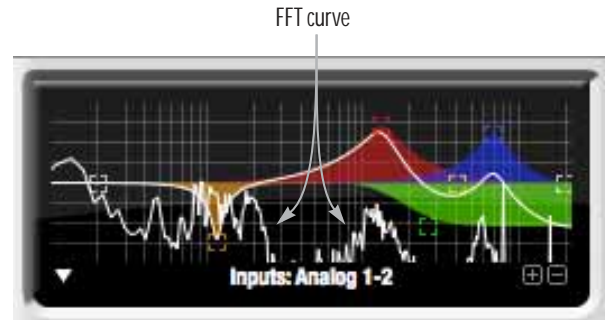
Figure 10-13: FFT display.

The FFT curve is post-filter. Therefore, the FFT shows the results of the EQ filter(s) being applied. Use the global EQ button for the input or output channel (Figure 10-3 and Figure 10-6, respectively) to toggle between the EQ'd and non-EQ'd FFT display.

Spectrogram

Choose *Show Spectrogram* from the *Filter display options* menu (Figure 10-10) to superimpose a real-time spectrogram "waterfall" display in the background of the EQ filter display, as demonstrated in Figure 10-14:
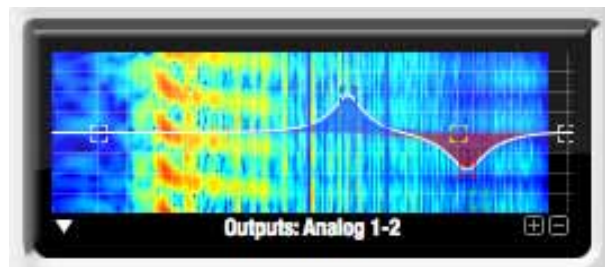
Figure 10-14: FFT display.

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

Figure 10-15: Spectrogram color-to-amplitude spectrum.

Using the full window filter display

Choose *Show Full Window Analysis* from the Filter display options menu (Figure 10-10) to fill the entire CueMix FX window with the filter EQ display for detailed inspection and adjustment of the EQ filter, as shown in Figure 10-16.

You can show and hide the FFT display, spectrogram or EQ band response curves as desired using the *Display Options* menu options (Figure 10-16). These settings are independent of the small graph display options (Figure 10-10), so you have the flexibility to display different combinations in each graph.

Show X-Axis Linear Scale

Check the *Show X-Axis Linear Scale* option in the Display Options menu (Figure 10-16) to display frequency on a linear scale, instead of the default logarithmic scale. With a linear scale (checked), frequency is constant, but the width of each octave along the x-axis is different. With a logarithmic scale (unchecked), octaves are displayed with a constant width, but frequency is displayed logarithmically within each octave.

The info box

The *Show Info Box* item in the full graph display options menu (Figure 10-16) lets you display the coordinates of any EQ filter point as you drag it in the graph (Figure 10-17):
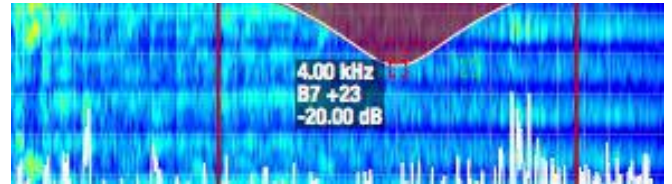
Figure 10-17: The Info Box.

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

Display options menu
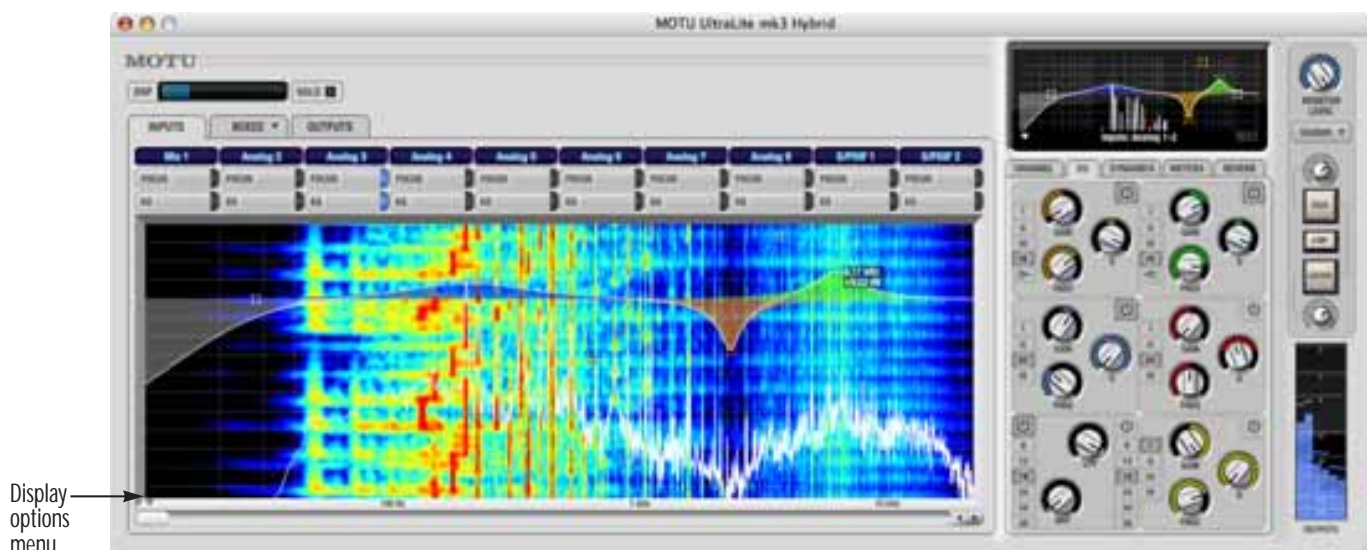
Figure 10-16: Full window filter display.

EQ filters

The EQ filters have three parameters:

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

Q

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

Filter types

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

Returning to zero (or nominal frequency)

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

EQ filter styles

EQ is one of the most widely used processing tools and can be applied to many different situations, from minor corrective tasks to highly creative applications. Over the years, many EQs have been engineered for specific applications or to achieve a certain sound. The Vintage EQ has been designed to be flexible enough to cover a broad range of applications. To that end, several different filter types are supplied, varying mostly in the way they handle the dynamic interaction between Gain and Q. This crucial relationship has been modeled to emulate the smooth and musical character of classic analog EQ circuits, in which the Gain/Q dependency was dictated by the actual circuit design and electrical components used.

The following sections describe the character of each type of EQ filter and their suggested applications. In the illustrations for each filter style (Figure 10-18 through Figure 10-21), the settings for the three example curves are the same for the purpose of comparison:

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

Type I
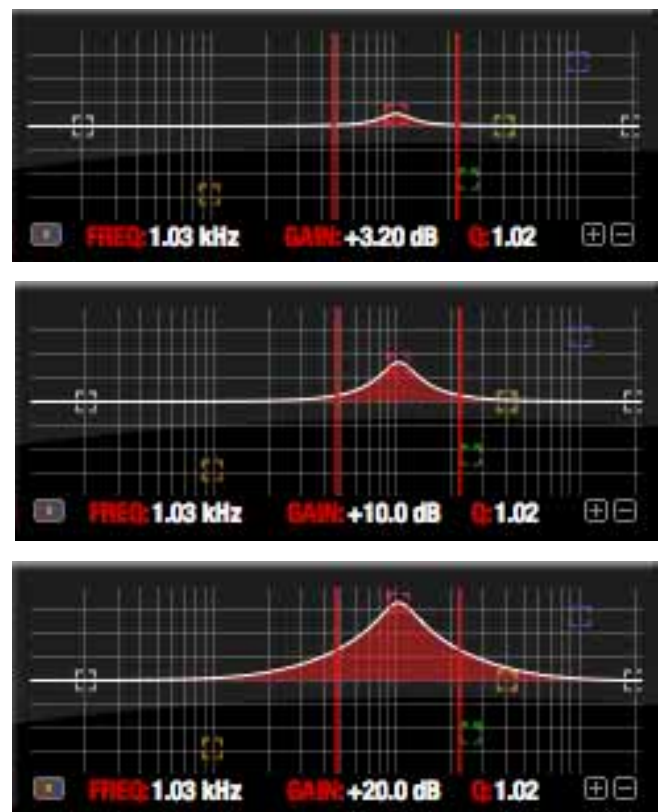
Figure 10-18: Type I EQ filter style.

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

Type II
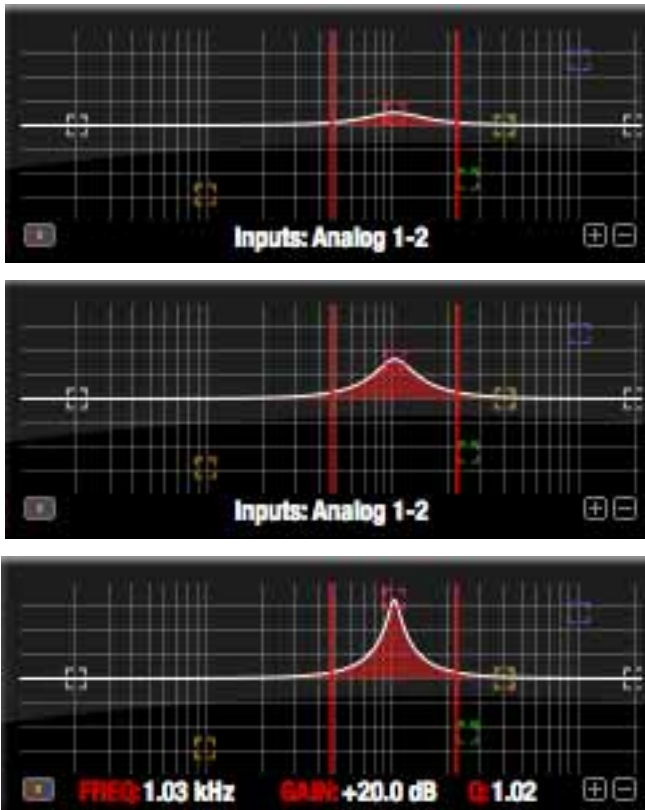
Figure 10-19: Type II EQ filter style.

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

Type III
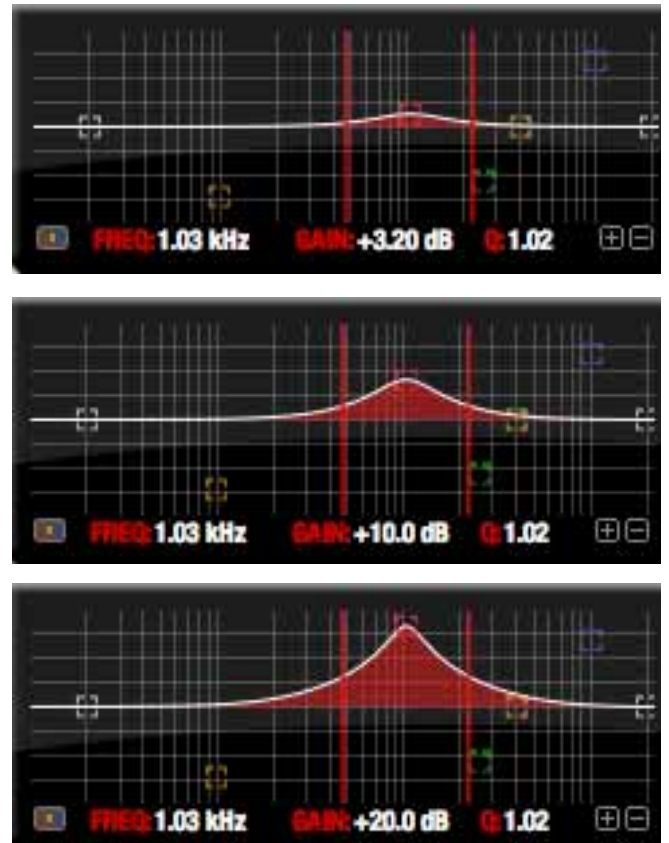
Figure 10-20: Type III EQ filter style.

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

Type IV
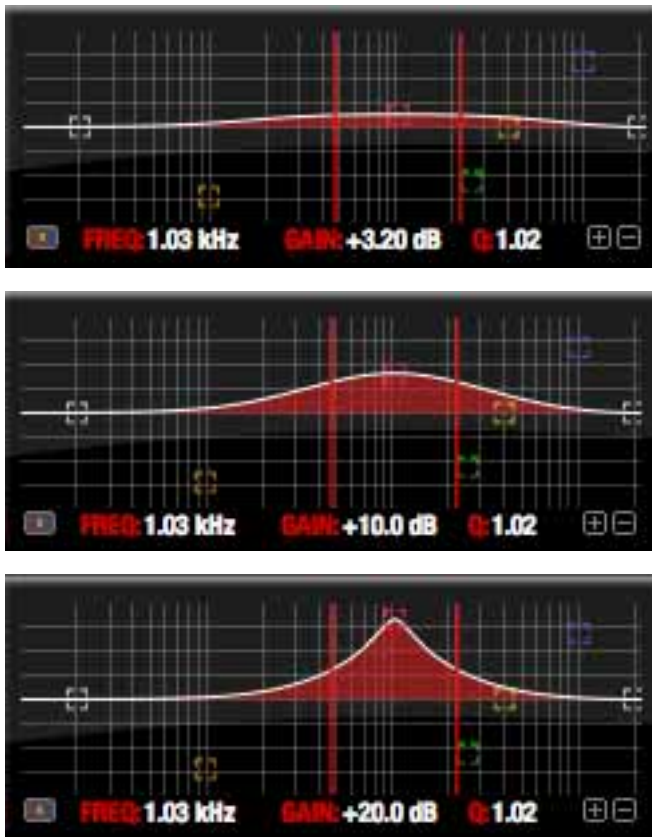
Figure 10-21: Type IV EQ filter style.

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

Shelf filters
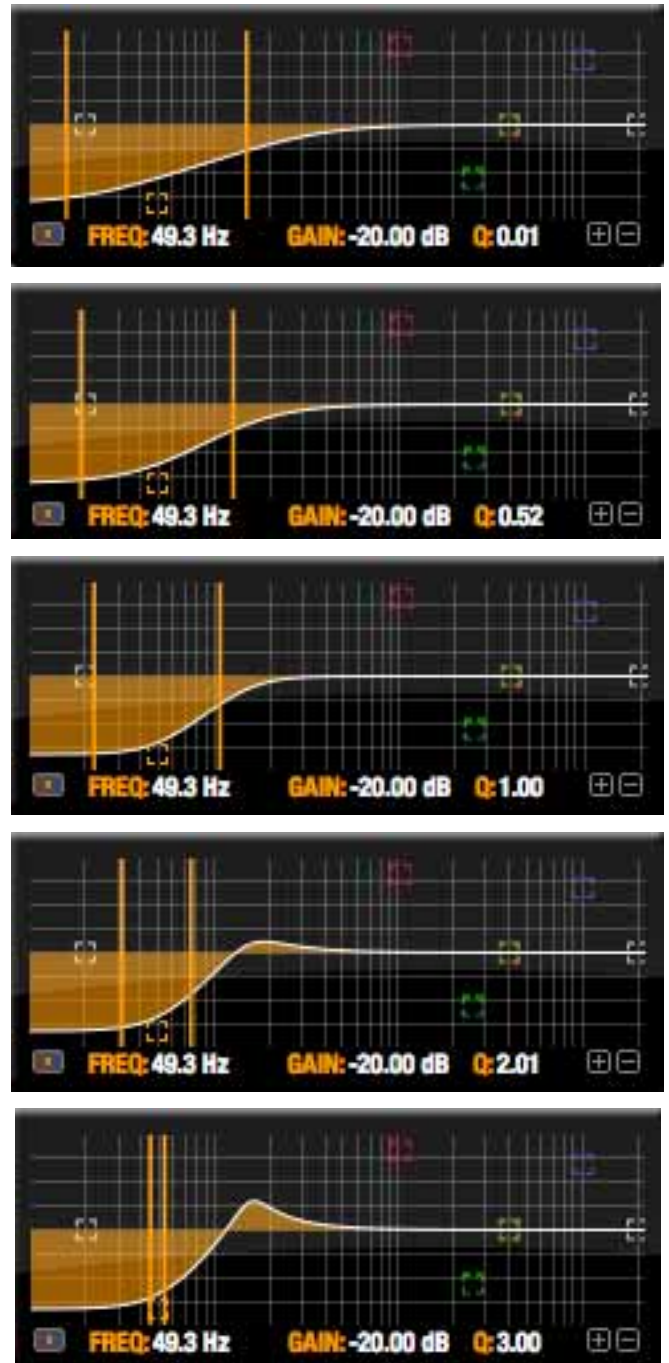
Figure 10-22: Shelf filter Q parameter overshoot.

When two top-most bands in the EQ tab are set to their *shelf filter* setting (Figure 10-10), the Q parameter controls the amount of overshoot applied to the response curve, as illustrated in Figure 10-22. 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 10-22 for Q values of 1.00, 2.00 and 3.00. This overshoot region produces a boost in frequencies just above the cutoff, which compensates in a smooth, more pleasing fashion for the perceived drop in low frequencies being cut.

Conversely, when shelving boost is being applied, overshoot cuts frequencies just above the cutoff to again compensate in a smooth and pleasing fashion for the perceived boost in low frequencies:
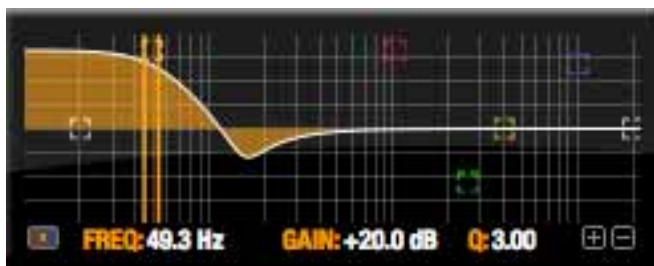
Figure 10-23: Overshoot when low shelf boost is applied.

Overshoot is also applied to high shelf boost and cut:
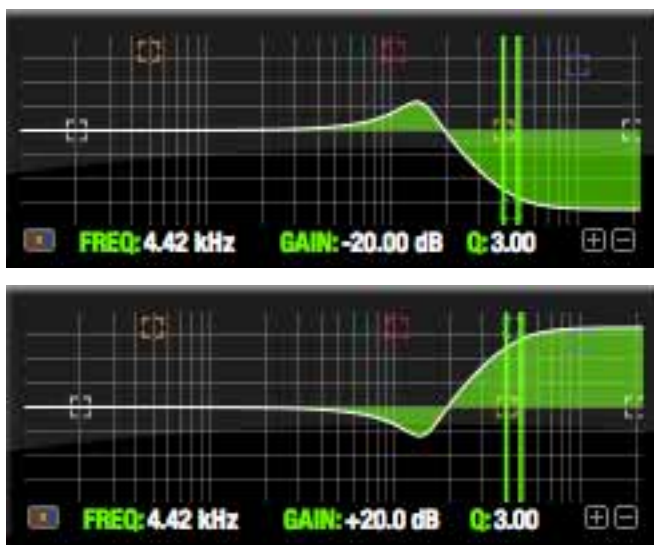
Figure 10-24: Overshoot when high shelf cut and boost is applied.

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

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

Overshoot curves are symmetrical for both cut and boost.

Low pass and high pass filters

The Vintage EQ low and high pass filters are similar to those found in most conventional parametric EQs (which usually have a fixed slope of 12 dB per octave), except that Vintage EQ provides six different slope (roll off) settings: 6, 12, 18, 24, 30 and 36 dB per octave. This control over the shape of the "knee" gives you a great deal flexibility and control for a wide variety of applications.
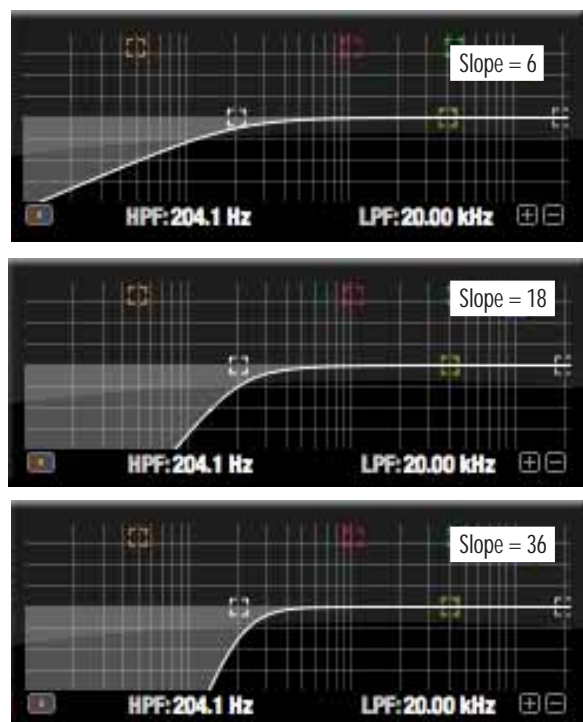
Figure 10-25: The low pass filter with three example slope settings.

The Dynamics tab

The Dynamics tab (Figure 10-26) displays the Dynamics processing settings for the input or output channel that currently has the focus. Click any focus button in the Inputs or Outputs tab to view the Dynamics tab settings for the channel.
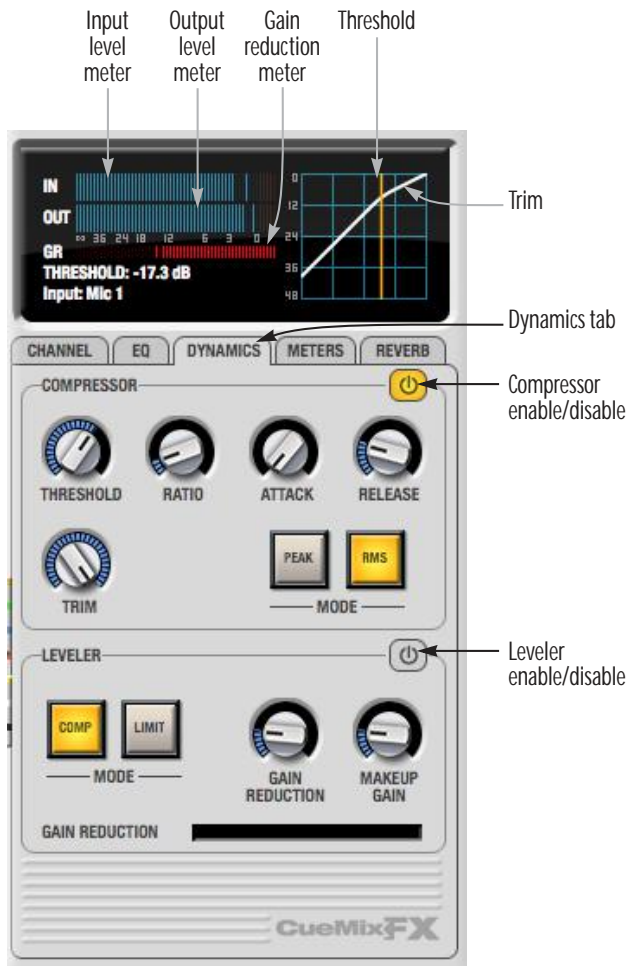
Figure 10-26: The Dynamics tab.

Enabling Dynamics

Each input and output channel has a global *Dynamics enable/disable* button (Figure 10-3 and Figure 10-6). 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 10-26) for the channel.

Compressor

The *Compressor* (Figure 10-26) 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 10-26).

Input level meter

The *Input Level* meter (Figure 10-26) 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 10-26) displays the current amount of attenuation applied by the compressor.

Output level

The *Output Level* meter (Figure 10-26) 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 10-26) 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 UltraLite-mk3 Leveler faithfully models the LA-2A using the on-board DSP with 32-bit floating point precision.

A model of an optical compressor

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

Automatic gain control using light

The the Automatic Gain Control (AGC) circuit of the LA-2A uses a vintage opto-coupler known by its model number (T4). The T4 contains an electroluminescent panel (ELP) and photoresistor mounted so that the emission of the panel modulates the resistance. An ELP consists of a thin layer of phosphorescent material sandwiched between two insulated electrodes to form a capacitor. Making one of the electrodes transparent allows the light to escape. These devices are essentially glow-in-the-dark paint on a piece of foil covered by metalized glass or plastic, and are the same devices used in low-power night lights. Unfortunately, these devices need high voltages to operate, and are best driven by tube circuits which can supply voltage swings of several hundred volts.

Response characteristics

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

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

Compressor/Limit buttons

The *Comp* and *Limit* buttons (Figure 10-26) 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 10-26) sets the strength of the signal sent to the AGC model.

Makeup Gain

Makeup gain (Figure 10-26) 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 10-27) serves as a comprehensive meter bridge for all inputs, outputs and mix busses in the UltraLite-mk3. This tab gives you a "bird's-eye" view of all signal activity in the UltraLite-mk3; it is ideal for confirming your signal routing programming and for troubleshooting.
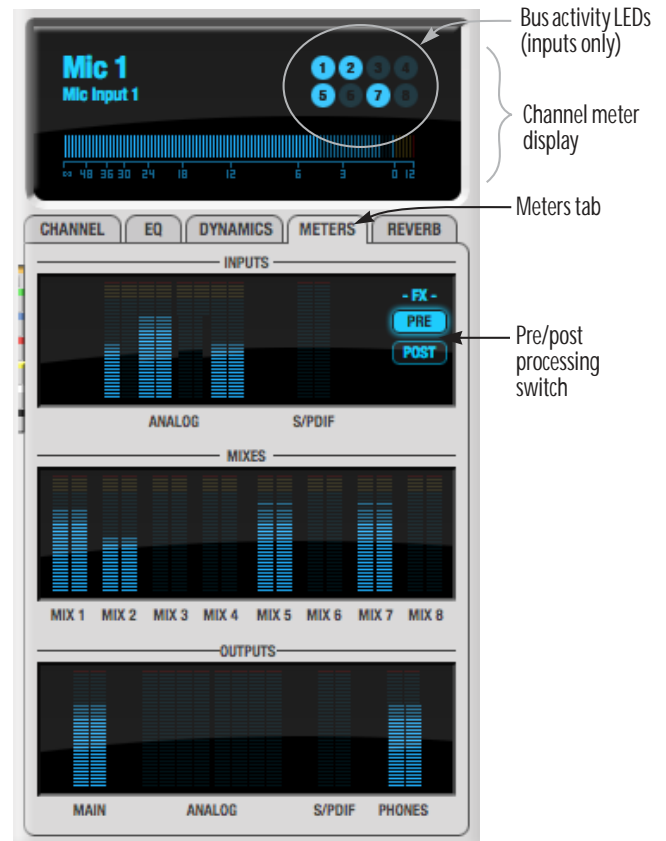
Figure 10-27: The Meters tab.

Channel meter display

The *channel meter display* (Figure 10-27) 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 10-27) are present only for inputs. See "Input meter and bus activity LEDs" on page 71.

Pre/post processing switch

The *pre/post processing* switch (Figure 10-27) 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 10-28) provides access to the UltraLite-mk3's single, global reverb processor, which provides high-fidelity reverberation and graphic control over its parameters.

Figure 10-28: The Reverb tab.

Enabling reverb

Use the enable/disable button (Figure 10-28) 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 10-3 on page 66)
- Mix bus output (Figure 10-2 on page 64)
- Outputs (Figure 10-6 on page 69)

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 10-28) 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 10-28) 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 10-28).

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

THE MONITOR GROUP
The *monitor group* is a set of UltraLite-mk3 outputs that can be controlled with the master Monitor Level knob in upper right corner of the CueMix FX mixer window (Figure 10-29), as well as the MASTER VOL knob on the UltraLite-mk3 front panel.
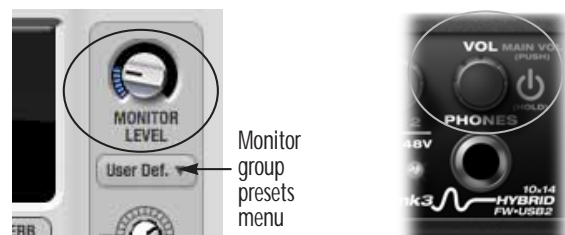
Figure 10-29: Monitor group volume control from CueMix FX and the front panel MASTER VOL knob.

Assigning outputs to the monitor group

Any combination of outputs can be assigned to the monitor group. To include an output pair in the monitor group, click its Monitor button in the Outputs tab (Figure 10-6 on page 69).

Monitor group presets menu

The *monitor group presets menu* (Figure 10-29) provides several presets for commonly used monitor groups:

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

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

Monitor group meters

The monitor group meters (Figure 10-1 on page 62) show levels for any/all output pairs that are currently included in the monitor group. The width of the meters scales proportionally so that all current monitor group outputs will fit within the prescribed space for the meters. If many outputs are included, then the meters will look fairly thin, but they will all be included in the meter.

DSP METER

The DSP meter (Figure 10-1) shows how much of the available DSP processing power is currently being used by the UltraLite-mk3 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 10-1) illuminates when any input in the current (active) mix bus is soloed (even if it is currently scrolled off-screen).

TALKBACK AND LISTENBACK

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

Hardware setup

Figure 10-30 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 UltraLite-mk3. For Listenback, set up a dedicated listenback mic in the live room for the musicians and connect it to another mic input. For talkback output, set up a headphone distribution amp or set of speakers in the live room, and connect it to a pair of outputs. For example, you might connect a headphone distribution amp to analog outputs 7-8 on your UltraLite-mk3, as demonstrated below in Figure 10-30.
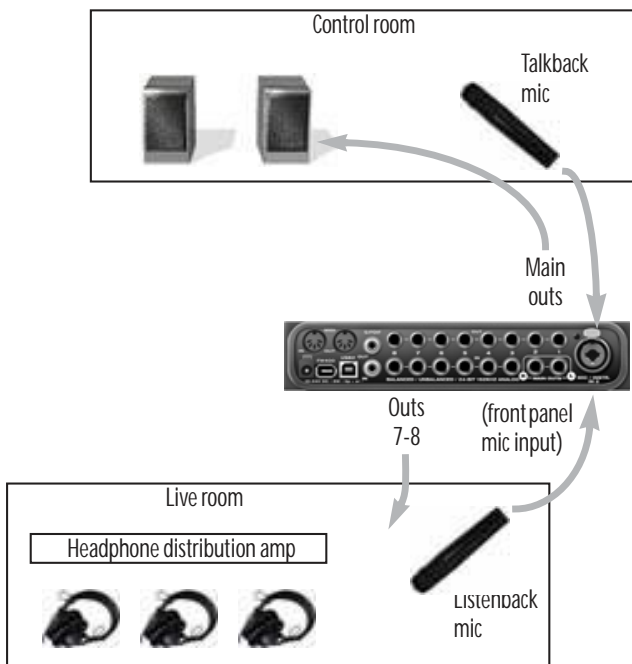
Figure 10-30: Typical hardware setup for Talkback and Listenback.

Talkback / Listenback Mic Input

To configure the talkback mic in CueMix FX, go to the Inputs tab (Figure 10-3 on page 66) and click the Focus button for the input that the talkback mic is connected to. Click the Channel tab (Figure 10-8 on page 70) 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 10-6 on page 69) 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 10-31) 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 10-31: The Talkback/Listenback controls.

Engaging/disengaging Talkback and Listenback

To engage Talk back or Listenback, press on the Talk or Listen buttons (Figure 10-31) and then release to disengage. Talkback and/or Listenback is engaged for as long as you hold down the mouse button. Alt-click to make the buttons "sticky" (stay engaged until you click them again — so you don't have to hold down the mouse). Or use the Talkback menu items.

If you would like to engage both Talkback and Listenback at the same time, enable the *Link* button (Figure 10-31).

Controlling Talkback and Listenback volume

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

SHORTCUTS

Hold down the following general modifier keys as shortcuts:

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

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

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

FILE MENU

Saving and loading hardware presets
The UltraLite-mk3 can store up to 16 presets in its on-board memory. A preset includes of all CueMix FX settings for all for 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 UltraLite-mk3.

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 UltraLite-mk3.

Hardware follows console stereo settings
This File menu item applies to other MOTU interfaces products and has no effect on the UltraLite-mk3.

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 10-1) and choose Copy from the Edit menu (or press control-C).

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

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

DEVICES MENU

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

Below each device are three signal analysis tools: Oscilloscope, X-Y Plot and Phase Analysis. Choose one to open its window. For details on these features, see the following sections.
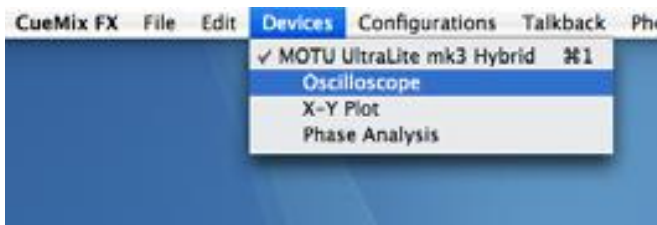
Figure 10-32: Opening the oscilloscope, X-Y Plot and Phase Analysis windows.

OSCILLOSCOPE

The Oscilloscope (Figure 10-33) 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 10-33, 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 mk3 series FireWire audio interface has its own oscilloscope. To open an oscilloscope, choose the *Oscilloscope* item from the Devices menu under the desired interface (Figure 10-32).

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 10-34) provide several options for the oscilloscope display.
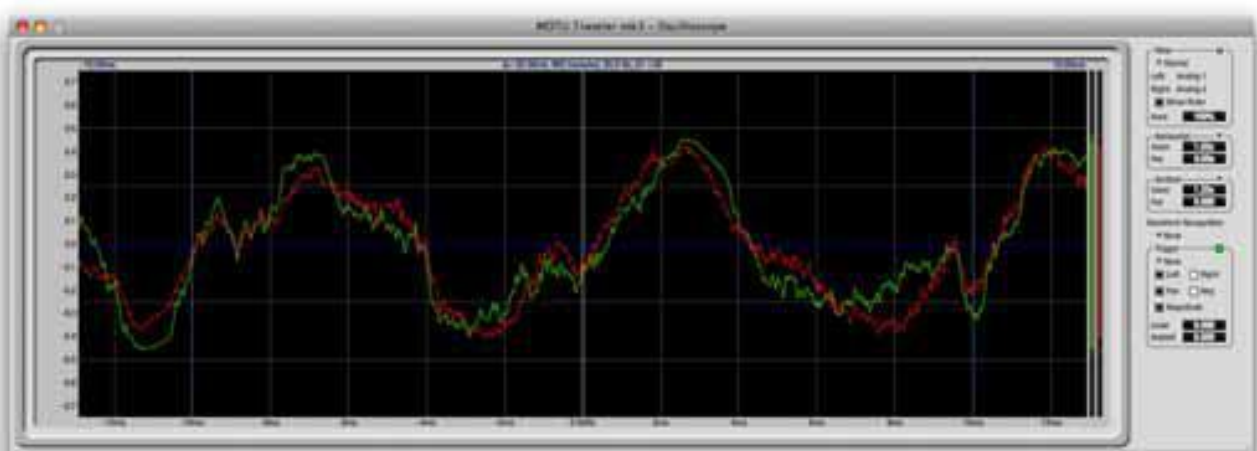
Figure 10-33: Oscilloscope
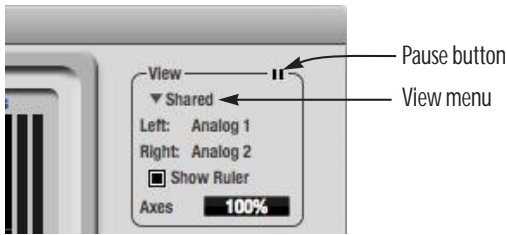
Figure 10-34: View controls

View menu

The View menu (Figure 10-34) lets you choose how you wish to display the audio channel(s) being displayed.

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

Display options

The *Axes* control (Figure 10-34) 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 91).

Pausing the display

The Pause button in the upper right corner of the View section (Figure 10-34) 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 10-35) 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 10-35).
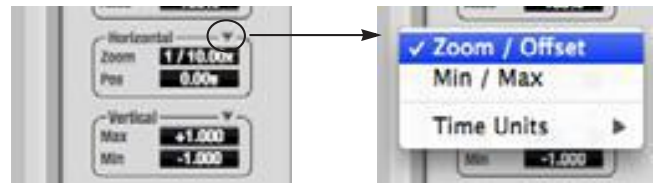
Figure 10-35: Horizontal control menu

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

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

Time Units

The *Time Units* sub-menu (Figure 10-35) provides the option to view the X axis in Seconds or Samples.

Vertical controls (amplitude axis)

The *Vertical* controls (Figure 10-35) operate similarly to the Horizontal controls, except that they configure the y-axis (amplitude).

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

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

Waveform Recognition

The Waveform Recognition option searches through new audio data looking for a waveform which most resembles that which was previously displayed. The region where this takes place is a small window around the line marking time equals zero, denoted by the extra vertical graph lines surrounding it. There are two kinds of waveform recognition available: Type I and Type II.
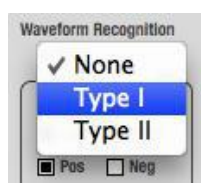
Figure 10-36: Waveform Recognition menu

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

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

Trigger

When the *Trigger* (Figure 10-37) is not enabled (the Trigger menu is set to *None*), the graph updates based on time: after every *n* samples of the monitored audio signal, the most recent samples are displayed. When the Trigger is enabled (set to any mode other than *None*), the graph updates in

response to specific conditions in the signal. The Trigger section defines that criteria and how the graph will display the events that match.
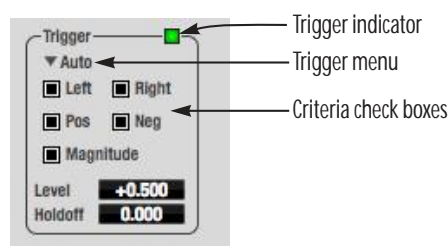
Figure 10-37: Trigger settings

Criteria

The criteria check boxes (Figure 10-37) determine the conditions that the trigger is looking for and where it will look for them.

The *Left* check box causes the condition to be looked for in the left channel of the signal; likewise, the *Right* check box 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* check boxes determine the slope of the event. When the *Pos* check box is enabled, the trigger will look for an event where amplitude is increasing; likewise, enabling the *Neg* check box 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* check box 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 10-37 on page 90) provides four modes:

Trigger mode	What it does
None	The Trigger is not active; this is the default mode. The incoming audio signal will be displayed continuously as audio is received.
Auto	The display is always updating, but when the condition is met, the trigger event will be displayed centered around the line marking time equals zero.
Normal	The display updates only when the condition is met; the last trace will be displayed until the next matching event is found.
Single Sweep	Similar to Normal mode, but the last trace will be displayed until you manually arm the trigger by clicking the Trigger indicator (Figure 10-37 on page 90) or by pressing the spacebar.

Trigger indicator

The Trigger indicator (Figure 10-37 on page 90) displays the state of the trigger, and also provides a way to manually interact with it. The Trigger indicator always displays one of three colors:

Color	Status
Green	When the current Trigger criteria has been met (including when the Trigger mode is *None*).
Yellow	When the Trigger is armed, but has not yet found an event which matches its criteria. Yellow can also indicate that the graph has been manually paused using the Pause button in the View section (see "Pausing the display" on page 89).
Red	When the Trigger is being held off, either because the Trigger mode is set to Single Sweep or the Holdoff time is not set to zero.

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

Measurement information

You can view detailed information about a particular time range by using the measurement bars.
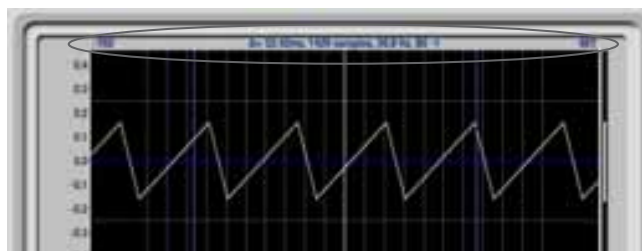
Figure 10-38: Measurement information

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

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

Ideas for using the Oscilloscope

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

Analyzing and comparing harmonic content

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

Viewing transients such as drum hits

If you loop a snare hit or other similar transient audio clip and feed it through the oscilloscope, you can more or less "freeze" the transient waveform in the oscilloscope frame. This can be useful, for example, for viewing the results of real-time compression that you are applying with an effects plug-in, as demonstrated in Figure 10-39. 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
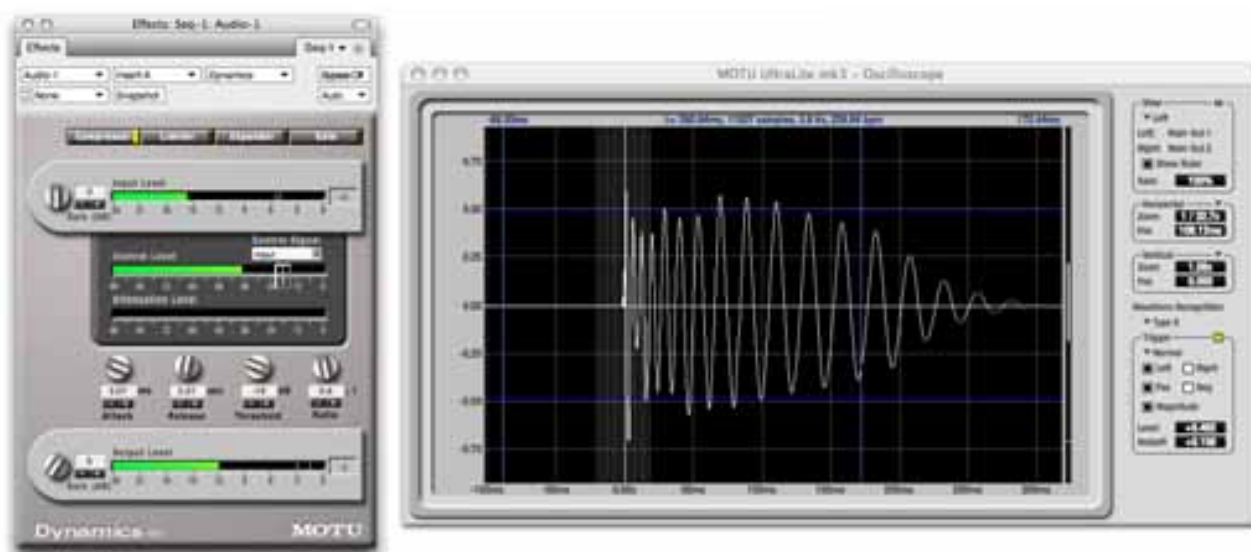
Figure 10-39: Viewing transients in the Oscilloscope

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 10-39.

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 10-37 on page 90), choose *Single Sweep* from the trigger menu (Figure 10-37), set the level to 0.999 and click the trigger indicator (Figure 10-37) 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 10-40) 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 10-40, below). There are also thick white diagonal lines for y = x and y = -x.

Metering

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

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

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

Opening the X-Y Plot

Each mk3 series FireWire audio interface has its own X-Y Plot window. Choose the *X-Y Plot* item from the Devices menu under the desired interface (Figure 10-32 on page 88).
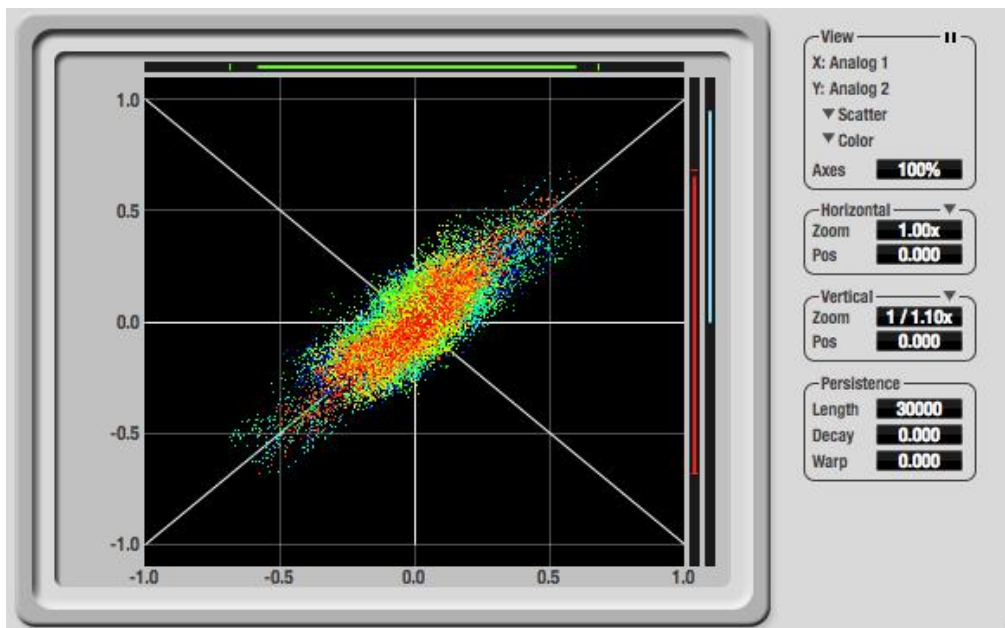
Figure 10-40: X-Y Plot

Choosing a channel pair to display

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

View controls

The View controls (Figure 10-41) provide several options for the X-Y Plot display.
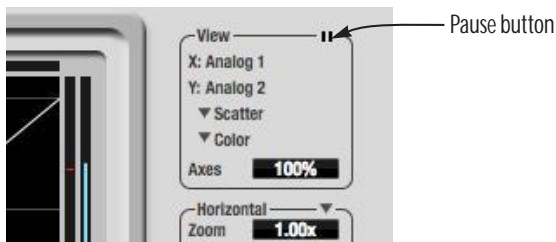
Figure 10-41: View controls

Pausing the display

The Pause button in the upper right corner of the View section (Figure 10-41) 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 10-41) 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 10-42.
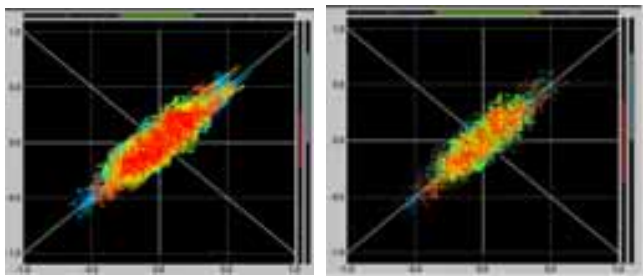
Figure 10-42: The same X-Y Plot displayed in Line versus Scatter mode.

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

Color/Grayscale

In *Color* mode (Figure 10-41) 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 96.

Axes

The *Axes* control (Figure 10-41) 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 10-43) 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 10-43.
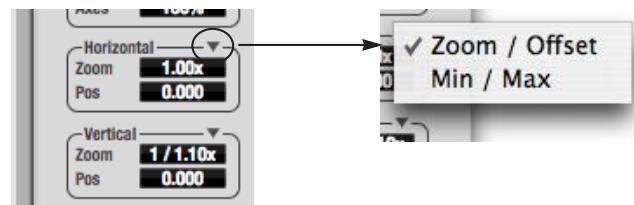
Figure 10-43: Setting the Horizontal or Vertical control modes.

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

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

Persistence

The *Persistence* controls (Figure 10-44) affect the appearance of data from when it is first displayed until it disappears from the grid.
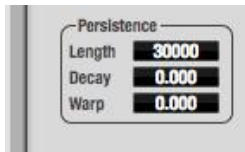
Figure 10-44: The Persistence controls.

Length

Length (Figure 10-44) 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 10-44 on page 96) 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 10-44) determines the position of data points after they are first drawn. When warp is zero, data points remain in the same position. When warp is positive, they contract towards the origin (center of the grid). When warp is negative, they expand away from the origin. The further the warp value is from zero, the greater the effect.

Using the X-Y Plot

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

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

If a stereo signal is out of phase, it is not mono compatible because it can cancel itself out, either partially or nearly completely, when collapsed to mono.
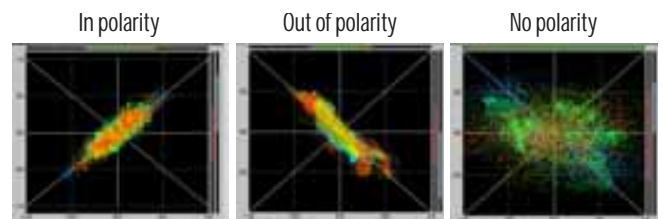
Figure 10-45: Checking polarity in a stereo signal with the X-Y Plot.

PHASE ANALYSIS

The *Phase Analysis* window (Figure 10-46 on page 97) graphs frequency versus phase difference versus amplitude of a stereo signal on either rectangular or polar coordinates.

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

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

Correlation Meter

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

Opening the Phase Analysis

Each mk3 series FireWire audio interface has its own Phase Analysis window. Choose the *Phase Analysis* item from the Devices menu under the desired interface (Figure 10-32 on page 88).

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 10-47) provide several options for the Phase Analysis display.
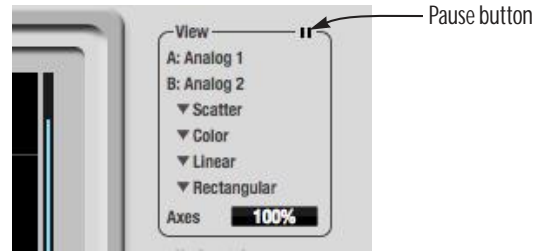
Figure 10-47: View controls

Pausing the display

The Pause button in the upper right corner of the View section (Figure 10-47) allows you to freeze the display at any time. To resume, click the button again. The correlation meter will remain active while the display is paused.
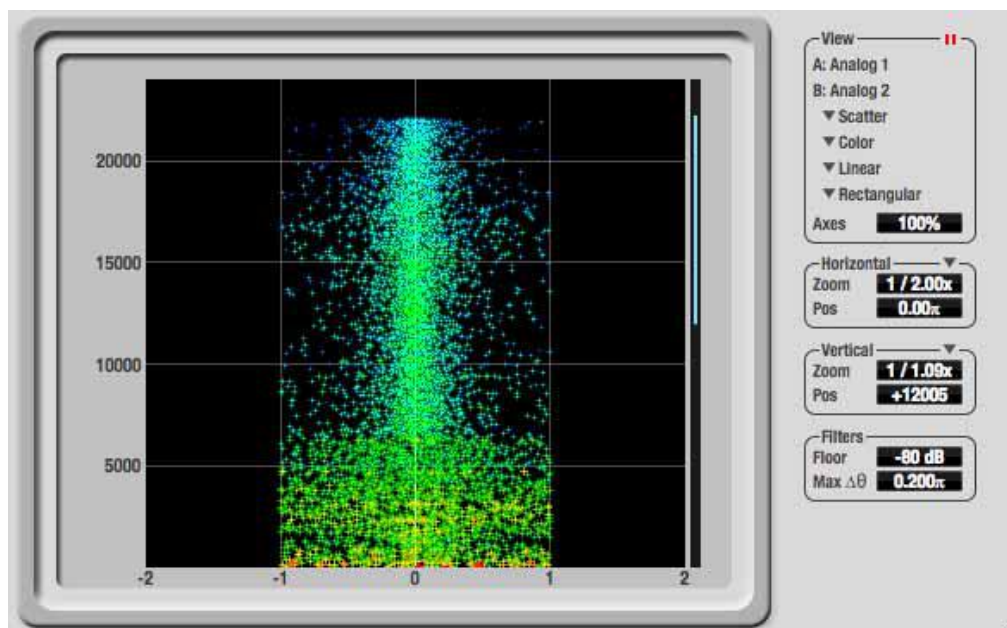
Figure 10-46: Phase Analysis

A/B (stereo audio channels)

The *View* section (Figure 10-47) 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 10-47) 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 10-42.
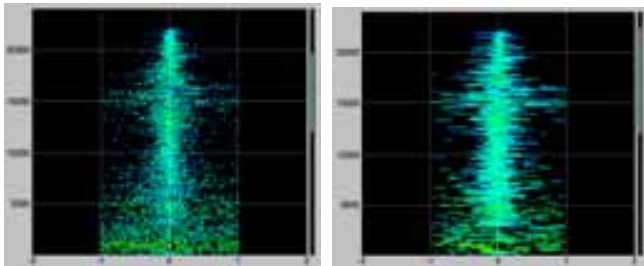
Figure 10-48: The same Phase Analysis displayed in Line versus Scatter mode.

☞ 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 99).

Color/Grayscale

In *Color* mode (Figure 10-47) 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 10-47) 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 10-47) 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.
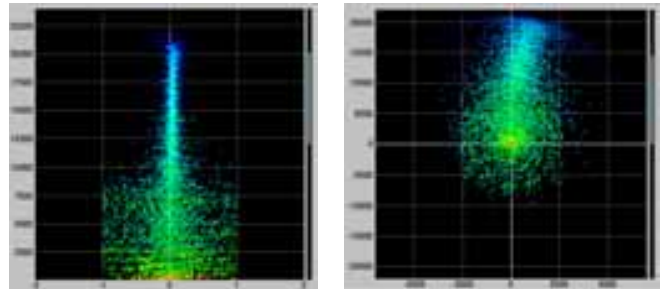
Figure 10-49: Rectangular versus Polar display (with a linear plot).

Above, Figure 10-49 shows Rectangular versus Polar display with a Linear plot. Below, Figure 10-50 show s the same displays (and the same data) with a Logarithmic plot:
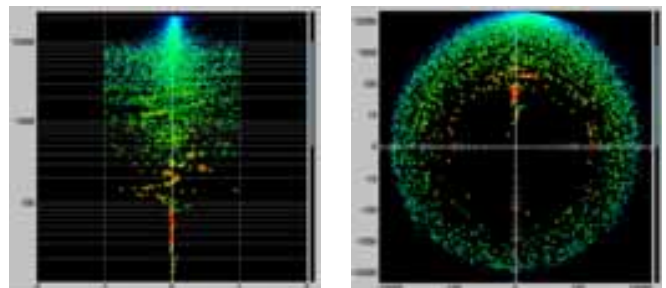
Figure 10-50: Rectangular versus Polar display with a logarithmic plot.

Axes

The *Axes* control (Figure 10-47) 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 10-51) 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 10-51.
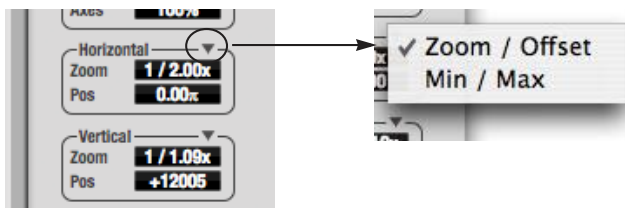
Figure 10-51: Setting the Horizontal or Vertical control modes.

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

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

Filters

The *Filters* section (Figure 10-52) lets you control the density of the Phase Analysis display.

Figure 10-52: Filters

Floor

Floor (Figure 10-52) 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 10-52) only affects Line view (see "Line/Scatter" on page 98) 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 10-53 on page 100), stereo material that is predominantly phase-aligned (correlated) appears along the vertical axis, as demonstrated in the first column (*Perfectly in phase*) in Figure 10-53. 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 10-53. Delays appear as spirals in the polar display.

The rectangular display (bottom row of Figure 10-53) 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 10-53 on page 100.

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 10-54 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).
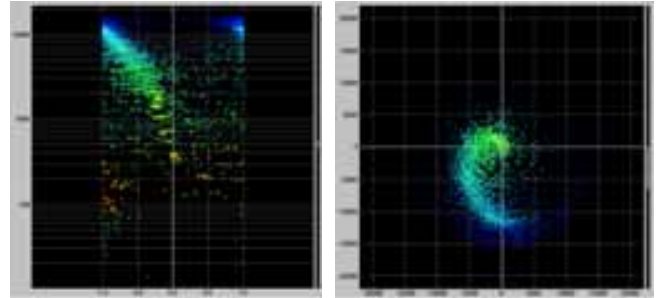
Figure 10-54: A stereo mix with phase issues.

CONFIGURATIONS MENU

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

	Perfectly in phase	One-sample delay	Twenty-sample delay	Inverted
Polar view				
Rectangular view				
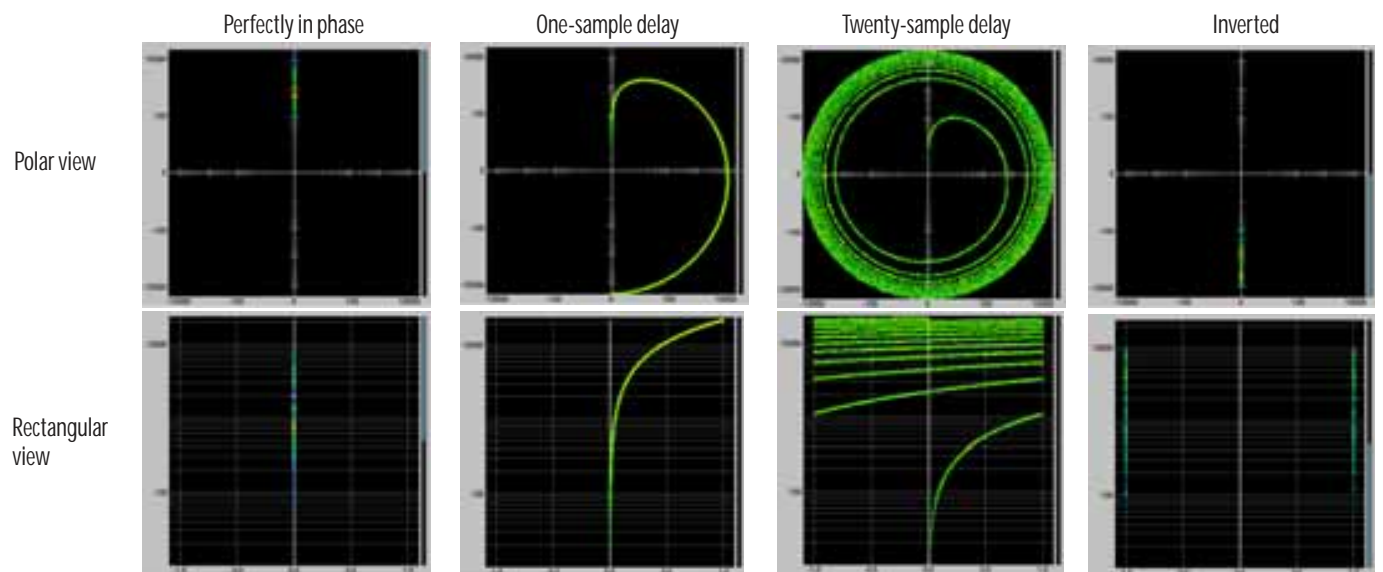
Figure 10-53: Two identical audio streams in the Phase Analysis.

Here is a summary of Configurations menu operations:

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

Modifying a configuration

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

Saving a CueMix FX configuration as a hardware preset

To save a CueMix FX configuration as a hardware preset:

1 Choose the configuration from the Configurations menu to make it the current active configuration.

2 Choose *File menu> Save Hardware Preset.*

3 Type in a name, choose a preset slot and click OK.

Saving a hardware preset as a CueMix FX configuration

To save a hardware preset as a CueMix FX configuration:

1 Choose *File menu> Load Hardware Preset* to make it the current active preset.

2 Choose *Configuration menu> Create New* (or *Save To*) to save it as a configuration.

TALKBACK MENU

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

PHONES MENU

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

CONTROL SURFACES MENU

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

Application follows control surface

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

that is not currently being displayed, CueMix FX will jump to the appropriate tab to display the control you are adjusting.

Share surfaces with other applications

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

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

CueMix Control Surfaces

CueMix FX includes support for the following control surface products:

- Mackie Control™

- Mackie HUI™

- Mackie Baby HUI™

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

Enabled

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

Configure…

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

Figure 10-55: Refer to the extensive on-line help for details about configuring CueMix FX for operation with your control surface product.

MOTU SMPTE Console

OVERVIEW

The UltraLite-mk3 can resolve directly to SMPTE time code via any analog input, without a separate synchronizer. The UltraLite-mk3 can also generate time code via any analog output. The UltraLite-mk3 provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy. Direct time code synchronization is supported by Cubase, Nuendo and other audio sequencer software that supports the ASIO2 sample-accurate sync protocol.

MOTU SMPTE 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 11-1) provides the same global *Clock Source* setting as in MOTU Audio Console ("Clock Source" on page 30), 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 UltraLite-mk3 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 UltraLite-mk3 can
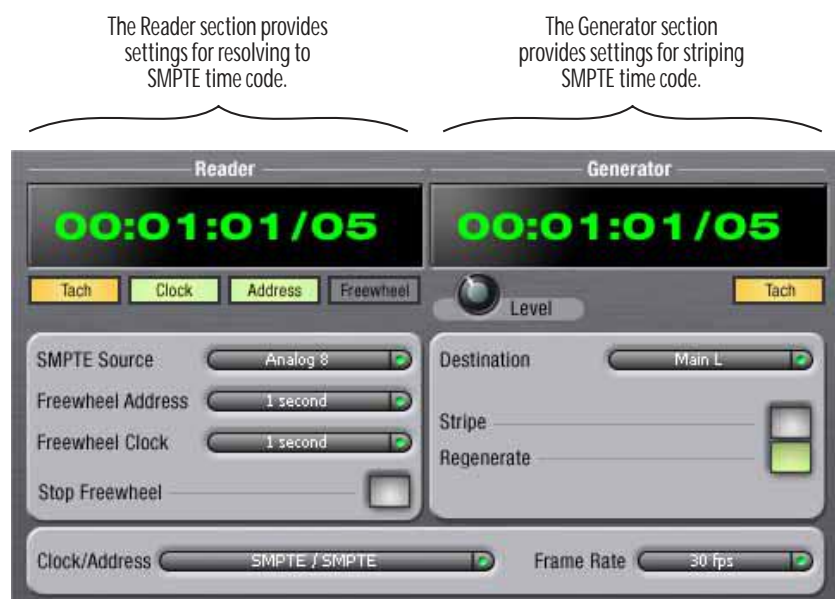
Figure 11-1: SMPTE Setup gives you access to your UltraLite-mk3's on-board SMPTE time code synchronization features.

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.

READER SECTION

The Reader section (on the left-hand side of the window in Figure 11-1) provides settings for synchronizing the UltraLite-mk3 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 UltraLite-mk3 has successfully achieved lockup to SMPTE time code and SMPTE frame locations are being read.

Clock

The Clock light glows continuously when the UltraLite-mk3 has successfully achieved lockup to an external time base, such as SMPTE time code or the S/PDIF input.

Address

The Address light glows continuously when the UltraLite-mk3 has successfully achieved lockup to SMPTE time code.

Freewheel

The Freewheel light illuminates when the UltraLite-mk3 is freewheeling address (time code), clock or both. For details about Freewheeling, see "Freewheel Address" and "Freewheel clock" below.

SMPTE source

Choose the analog input that is connected to the time code source. This is the input that the UltraLite-mk3 "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 UltraLite-mk3 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 UltraLite-mk3 to freewheel before it gives up and stops altogether.

The UltraLite-mk3 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 UltraLite-mk3 to freewheel indefinitely, until it receives readable time code again. To make it stop, click the *Stop Freewheeling* button.

Freewheel clock

Freewheeling occurs when there is glitch or drop-out in the incoming SMPTE time code for some reason. The UltraLite-mk3 can freewheel past the drop-out and then resume lockup again as soon as it receives a stable, readable clock signal.

The UltraLite-mk3 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 UltraLite-mk3 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 11-1) 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 UltraLite-mk3. The level knob disappears when the Destination is set to *None*.

Tach light
The Tach light blinks once per second when the UltraLite-mk3 is generating SMPTE time code.

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

Stripe
Click this button to start or stop time code. To set the start time, click directly on the SMPTE time code display in the Generator section and type in the desired start time. Or drag vertically on the numbers.
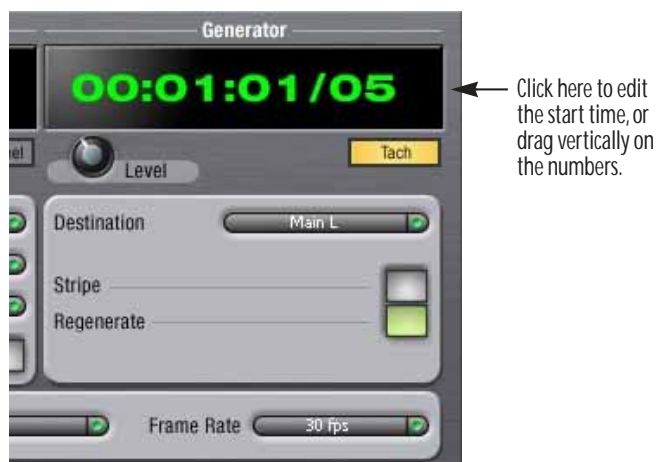
Figure 11-2: Setting the time code start time.

Click here to edit the start time, or drag vertically on the numbers.

Regenerate
This option, when enabled, causes the generator to generate time code whenever the UltraLite-mk3 is receiving SMPTE time code.

RESOLVING CUBASE OR NUENDO TO SMPTE TIME CODE
To resolve your UltraLite-mk3 and Cubase or Nuendo directly to SMPTE time code with no additional synchronization devices, use the setup shown in Figure 11-3 on page 106. Make sure the *Clock Source* setting in the MOTU Audio Console window is set to *SMPTE*. Also, make sure that you've connected an LTC input signal to an UltraLite-mk3 analog input, and that you've specified that input in the SMPTE Setup.

SYNCING TO SMPTE TIME CODE

The UltraLite-mk3 system can resolve directly to SMPTE time code. It can also generate time code, under its own clock or while slaving to time code. Therefore, the UltraLite-mk3 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 UltraLite-mk3 to slave your audio software to time code as well, as long as your software supports sample-accurate sync, which is the means by which the software follows the UltraLite-mk3. The accuracy may not be sample-accurate, but in most cases it will be very close.

Use this setup if you have:

✓ A SMPTE time code source, such as a multitrack tape deck.

✓ An UltraLite-mk3 by itself, OR with another slaved device (such as a digital mixer).

✓ Host software that supports sample-accurate sync.

This setup provides:

✓ Continuous sync to SMPTE time code.

✓ Sub-frame timing accuracy.

✓ Transport control from the SMPTE time code source.

SMPTE time code source

audio cable bearing LTC (Longitudinal Time Code)

Any analog input quarter-inch jack

UltraLite-mk3 interface

FireWire cable

Windows computer running Cubase or other sample-accurate ASIO software
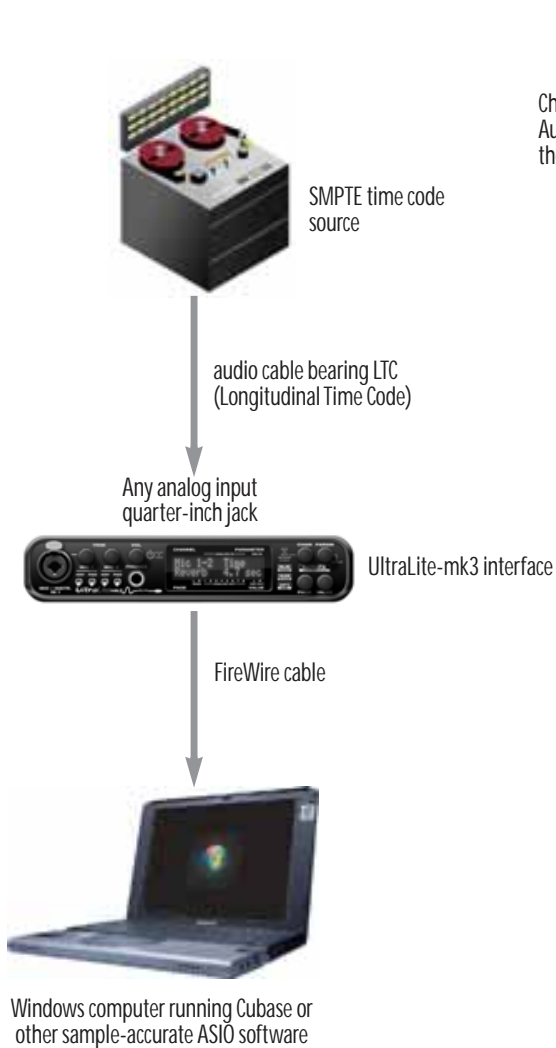
Choose *SMPTE* as the clock source in MOTU Audio Setup. This setting can also be made in the MOTU SMPTE Console (shown below).
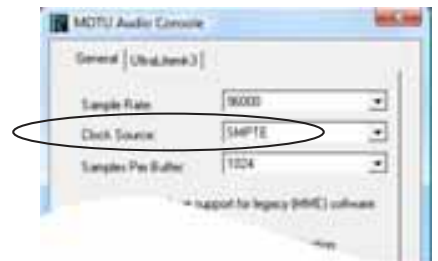
In Cubase, go to the Synchronization window and choose *ASIO Audio Device* as the Timecode Base.

Launch the MOTU SMPTE Console and specify the *SMPTE Source*, which is the interface receiving the SMPTE time code. Also, confirm that the *Clock Source/Address* is *SMPTE/SMPTE*.

Figure 11-3: Connections for synchronizing the UltraLite-mk3 directly to SMPTE time code.

CHAPTER 12 Performance Tips & Troubleshooting

Things seems like they are working fine, but the UltraLite-mk3 then just drops off line, and the computer can't see it anymore on the FireWire bus.
This is a common symptom when the problem is that the UltraLite-mk3 is not getting enough power. Check the power source for the UltraLite-mk3. Make sure that the power connection meets the requirements outlined in "Bus power requirements" on page 23.

Why does the start sound not play through the UltraLite-mk3?
The UltraLite-mk3 will only playback audio at a sampling rate of 44.1, 48, 88.2. 96. 176.4 or 192 kHz.

The computer freezes when it starts up
If the computer is unable to boot up, it may be a conflict with the WDM Driver. As soon as the computer boots up, Windows will try and initialize the WDM Driver. If this fails, your computer will hang. To determine if the WDM Driver is the problem, boot up in Safe Mode or remove the MOTU Audio software with the Add/Remove Programs Control Panel then restart. Reinstall the UltraLite-mk3 software and choose only to install the MOTU ASIO driver, not the WDM Driver. Restart again.

'New hardware detected' window
If you connect your UltraLite-mk3 before running the UltraLite-mk3 Software Installer CD, Windows will prompt you that new hardware has been detected. Cancel this, and run the UltraLite-mk3 Software Installer, rather than allowing Windows to locate the drivers.

UltraLite-mk3 inputs and outputs are not available in SONAR
Make sure that the inputs and/or outputs that you want to use are enabled in SONAR.

Sound Forge and 'Smooth Scroll'
If the Smooth Scroll option under the Options menu is turned on, you may experience clicks and pops or other anomalies in the audio playback. This gets worse as you zoom in more to the waveform while playing. For the best quality playback, turn this option off.

Clicks and pops due to hard drive problems
If you have checked your clock settings sync cable connections and you are still getting clicks and pops in your audio, you may have a drive related problem. Set the UltraLite-mk3 Clock Source to Internal and try recording just using the analog inputs and outputs of the UltraLite-mk3. 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 the disk drivers are outdated.

If you are using a FireWire drive on the same bus as the UltraLite-mk3, it could be that the FireWire bus is overloaded (too many devices on the same bus). Try removing all devices except the UltraLite-mk3.

Connecting or powering gear during operation
It is not recommended that you connect/disconnect, or power on/off devices connected to the UltraLite-mk3 while recording or playing back audio. Doing so may cause a brief glitch in the audio.

Monitoring - how do I 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 UltraLite-mk3's hardware-based

CueMix FX monitoring feature. Please see chapter 9, "Reducing Monitoring Latency" (page 55).

Controlling monitoring latency
See chapter 9, "Reducing Monitoring Latency" (page 55).

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 on line at motu.com, or fill out and mail the registration card included with your 8pre. Doing so entitles you to technical support and notices about new products and software updates.

REPLACING DISKS

If your UltraLite-mk3 installer CD becomes damaged and fails to provide you with fresh, working copies of the program, 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 UltraLite-mk3 system, you may contact our technical support department in one of the following ways:

■ Tech support hotline: (617) 576-3066 (Monday through Friday, 9 am to 6 pm EST)

■ Tech support 24-hour fax line: (617) 354-3068

■ 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 UltraLite-mk3 system. This is printed on a sticker placed on the bottom of the UltraLite-mk3 itself. 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 UltraLite-mk3 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 UltraLite-mk3 Development Team, MOTU Inc., 1280 Massachusetts Avenue, Cambridge, MA 02138.

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-10dB analog input 20
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