

MicroBook II CueMix FX Guide

OVERVIEW

CueMix FX is a cross-platform software application that provides graphic, on-screen control for the MicroBook II's flexible CueMix FX on-board mixer and effects processing. CueMix FX can be used independently of host audio software, or together with it.

- An 8-bus mixer with EQ and compression 1
- CueMix FX versus host mixing 1
- CueMix FX installation 1
- Opening CueMix FX 1
- CueMix FX basic operation 2
- Inputs tab 3
- Mixes tab 5
- Outputs tab 7
- Channel settings section 8
- Meters tab 15
- Signal Generator tab 16
- Output volume controls 16
- Device Setup 16
- FFT and Spectrogram display 17
- Oscilloscope 20
- X-Y Plot 24
- Phase Analysis 26
- Tuner 30
- Shortcuts 30
- File menu 31
- Edit menu 31
- Devices menu 31
- Configurations menu 31
- Phones menu 32
- Control Surfaces menu 32

AN 8-BUS MIXER WITH EQ AND COMPRESSION

All MicroBook II inputs and outputs can be mixed and controlled by the on-board CueMix FX 8-bus (4 stereo bus) digital mixer driven by hardware-based DSP.

The CueMix FX mixer allows you to apply no-latency EQ and dynamics to inputs and outputs directly in the MicroBook II hardware, independent of the computer. Input signals to the computer can be recorded wet, dry, or dry with a wet monitor mix (for musicians during recording, for example).

Effects include:

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

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

The MicroBook II's flexible effects architecture allows you to apply seven bands of EQ and compression on every input and output simultaneously.

CUEMIX FX VERSUS HOST MIXING

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

- CueMix FX has no buffer latency. Thanks to the MicroBook II's DSP chip, CueMix FX provides the same throughput performance as a digital mixer.
- CueMix FX mixing and effects processing imposes no processor drain on the computer's CPU.
- CueMix FX routing can be maintained independently of individual software applications or projects.

CUEMIX FX INSTALLATION

CueMix FX is installed with the rest of your MicroBook II software.

OPENING CUEMIX FX

When CueMix FX is opened for the first time, you will see a prompt with your MicroBook II serial number advising you to register your MicroBook II. While registration is not necessary to use the MicroBook II, it is required for technical support and software updates.

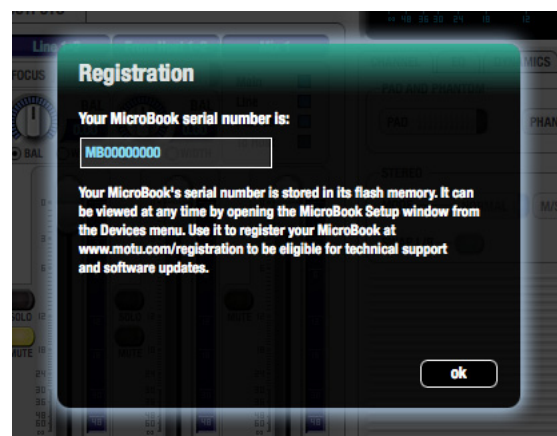


Figure 1: Launching CueMix FX for the first time

CUEMIX FX BASIC OPERATION

Here is a brief overview of the CueMix FX mixer.

Four stereo mix busses

CueMix FX provides four stereo mix busses: Mix 1, Mix 2, Mix 3, and Mix 4. Each mix bus can take any number of inputs and mix them together into a single stereo audio stream that you can then assign to any MicroBook II output pair — or combination of output pairs — that you choose. For example, Mix 1 could go to the headphones, Mix 2 could go to the main outs, Mix 3 could go to a piece of outboard gear connected to analog outputs 3-4, and so on. You can also assign a mix to multiple output pairs. For example, you could assign Mix 1 to the Main Out, Phones, and Line Out.

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 2 on page 2). To select which mix you are viewing, choose it from the mix bus menu. The mix name appears above the mix bus master fader, 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 3 on page 3) gives you access to settings for individual MicroBook II inputs (or input pairs), such as phase, trim, EQ and dynamics processing. 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 7 on page 7) gives you access to settings for each MicroBook II output pair, including EQ and dynamics processing. These settings are applied to the signal just before it is sent to the output.

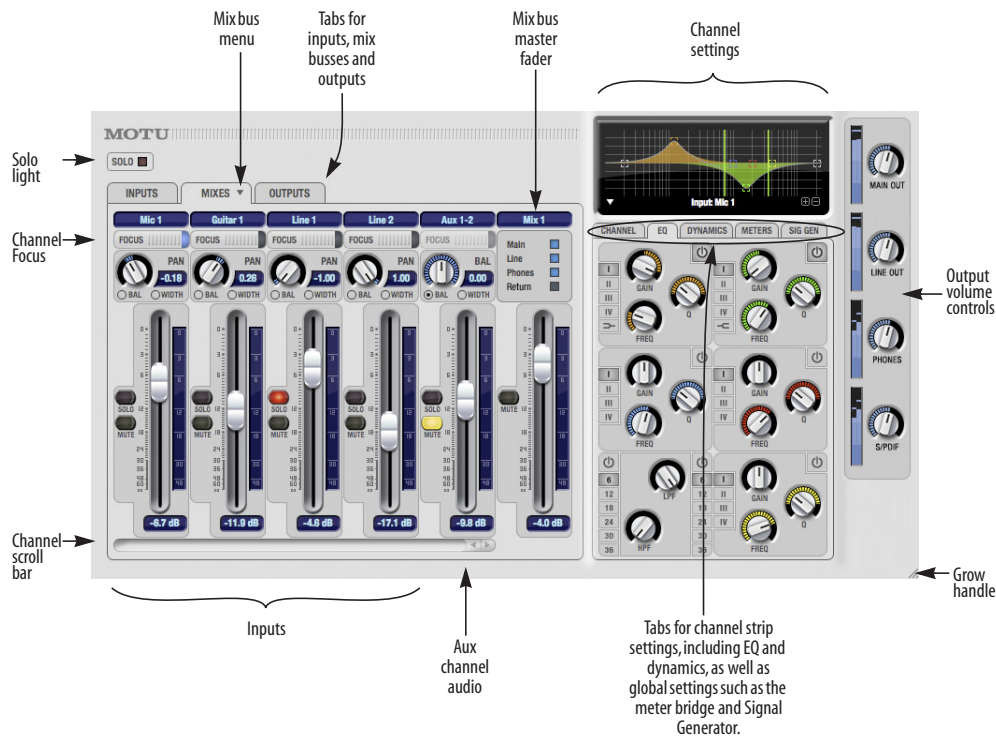


Figure 2: CueMix FX is a virtual mixer that gives you control over the MicroBook II's on-board mixing features.

Channel focus and settings

Click the focus button for a channel (Figure 2) to view channel-specific parameters in the Channel Settings section of the CueMix FX window (Figure 2). Separate tabs are provided for channel-specific settings (channel strip, EQ and dynamics), plus the global meter bridge and signal generator.

Output volume controls

The *Monitoring section* (Figure 2) provides independent volume control for all of the MicroBook II's outputs.

Other features

CueMix FX offers many additional features, such as extensive metering, graphic editing of certain effects parameters, a signal generator, and more.

Widening the CueMix FX window

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

INPUTS TAB

The MicroBook II provides many features for managing input signals. Some of these features, such as the MicroBook II'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 3) to access and control all of these input channel settings for each MicroBook II input or input pair.

Input tab settings are global

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 (via the MicroBook II's *Return 1-2* channels in your host), do not apply any changes to it in the Input tab, or change the *Scope/Return Channel Selection* to *Pre-Effects* (see "Device Setup" on page 16).

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, and dynamics.

Input channel focus

Click the channel focus button (Figure 3) to view and edit parameters in the channel settings section of the window (Figure 8 on page 8), including the Channel, EQ, and Dynamics tab settings for each input.

Mono/stereo pairing

Click the *Mono* button (Figure 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*

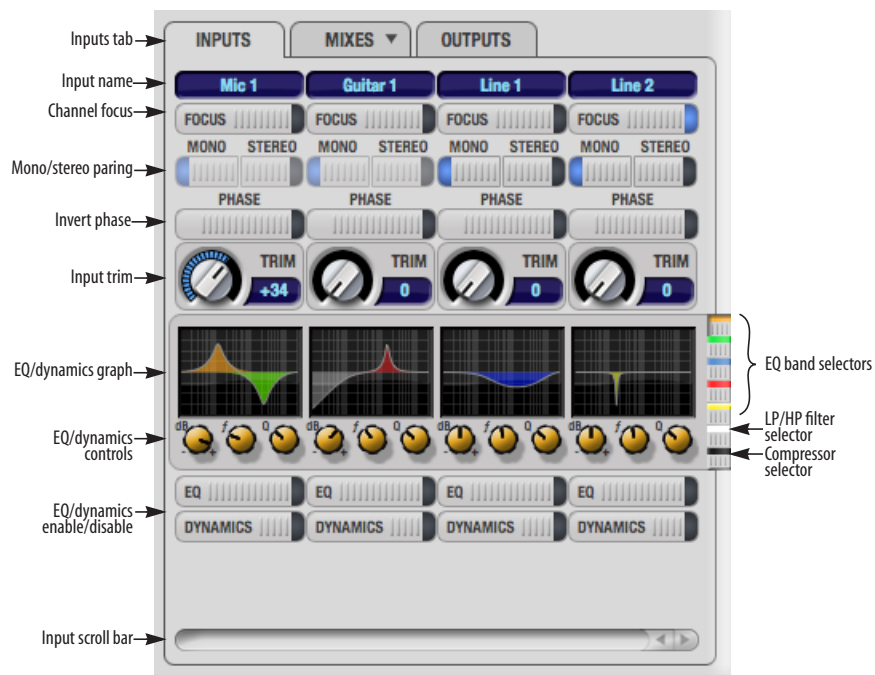


Figure 3: The Inputs tab.

button. Inputs are grouped in odd/even pairs (Line 1-2, for example). Stereo pairs appear as a single channel strip in the CueMix FX mixer (in all tabs).

Invert phase

The *Phase* button (Figure 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 MicroBook II inputs offer continuously variable input trim. Here is a summary of input trim ranges for each type of MicroBook II input:

Input	Trim cut	Trim boost	Trim Range
Mic	0 dB	51 dB	51 dB
Guitar	0 dB	32 dB	32 dB
TRS analog inputs	-96 dB	0 dB	96 dB

Input EQ and dynamics

The MicroBook II lets you apply 7-band parametric EQ and dynamics processing (DSP) to any input.

The controls in the EQ/Compression section of the Inputs tab (Figure 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 11 on page 9). 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 “Channel settings section” on page 8.

The EQ/Dynamics graph

The EQ/Dynamics graph for each input channel strip (Figure 4) provides a thumbnail view of the EQ curves or Compressor graph for the channel. To change the EQ settings in this graph, use the two or three knobs below, as explained in the following sections. The EQ graphic is for display purposes only; it cannot be edited directly. If you would like to edit the EQ curves graphically, you can do so in the EQ tab (Figure 11 on page 9). When viewing the Compressor graph, you can drag the orange bar to adjust the compressor threshold.

EQ/Dynamics selectors

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

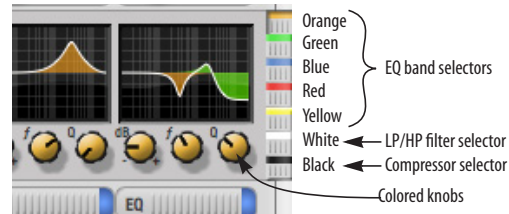


Figure 4: The EQ/Dynamics selectors.

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

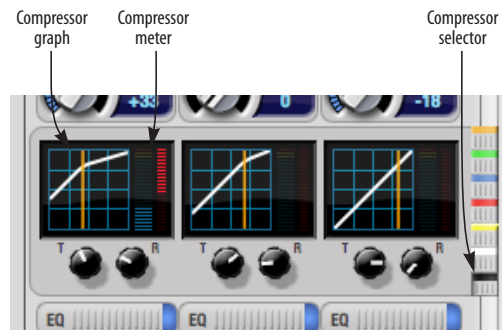


Figure 5: The Compressor controls.

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

Using the EQ/Dynamics knobs

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

🖱️ **Important:** before you can modify the settings of an EQ band using the three knobs below the graph, the EQ band must be enabled. This is done in the EQ tab (Figure 11), as explained in “Enabling EQ” on page 9.

EQ/Dynamics enable/disable buttons

Click the EQ or Dynamics button at the bottom of the input channel (Figure 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.)

MIXES TAB

Click the *Mixes* tab (Figure 6) to gain access to the MicroBook II's four stereo mix busses. The Mixes tab displays one mix bus at a time: choose each mix from the *Mix bus menu* in the tab itself. Each fader represents an input to the mix, except the fader on the right, which is the master fader for the mix. To assign each mix to a MicroBook II output, visit the Outputs tab.

Input section

The horizontally scrolling area in the Mix tab to the left of the master fader (Figure 6) displays channel strips for all MicroBook II inputs.

Input channel focus

Click the channel focus button (Figure 6) to view and edit parameters in the channel settings section of the CueMix FX window (Figure 8 on page 8).

Input pan section

The input pan knob (Figure 6) 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 and Width.

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.

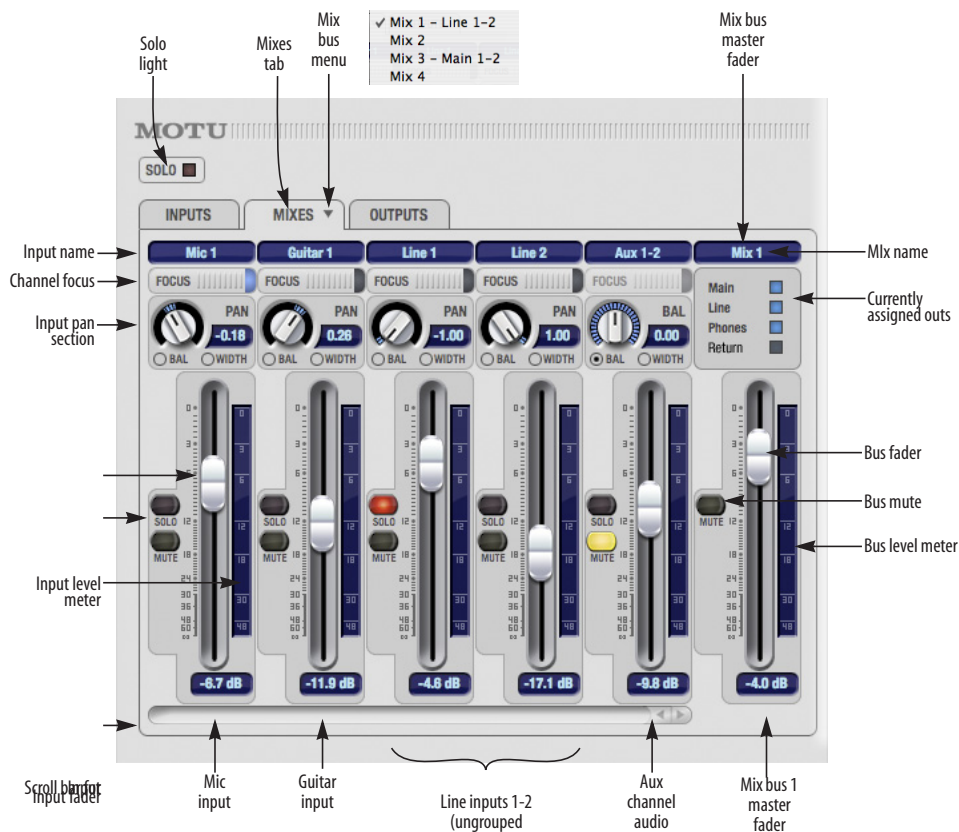


Figure 6: The Mixes tab.

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 6 on page 5) 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 2) will illuminate.

Aux channel

Inputs consist of physical inputs on the MicroBook II interface (mic, guitar, etc.), but there is also a stereo input channel for *Aux 1-2*. This is audio being routed to the MicroBook II from your host audio software. By bringing up this fader, you can combine live inputs (mic, guitar, and line 1-2) with any audio from the computer.

Stereo grouping

The input channel strips in the Mixes tab can represent either a mono input channel or a stereo pair. This is controlled by their grouping in the Inputs tab. So if you want to change the configuration from mono to stereo, or vice versa, do so using the Mono/Stereo buttons in the Inputs tab (Figure 3 on page 3).

Solo light

The Solo light (Figure 6 on page 5) illuminates when any input in the current (active) mix bus is soloed, even if it is currently scrolled off-screen.

Naming a mix

Click the mix name at the top of the mix bus master fader (Figure 6) to edit the name.

Mix bus output assignments

The outputs currently assigned to a mix are displayed above the mix's master fader. For details on choosing the source for an output, see "Output source" on page 7.

Bus master fader

The bus fader (Figure 6 on page 5) controls the overall level of the mix (its volume on its stereo output). Use the individual input faders to the left to control individual input levels.

Bus master mute

The bus mute button (Figure 6 on page 5) disables (silences) the mix.

Bus level meter

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

OUTPUTS TAB

The *Outputs* tab (Figure 7) lets you apply EQ and dynamics to any output pair, just before the signal leaves the MicroBook II. This is processing that occurs at the very end of the signal flow, after everything else (host based effects, MicroBook II 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.

Output channel focus

Click the channel focus button (Figure 7) to view and edit parameters, such as EQ and dynamics, in the Output Settings section of the CueMix FX window.

Output EQ and Dynamics

The EQ/Dynamics section in the Outputs tab (Figure 7) works identically to the EQ/Dynamics section for the Inputs tab (Figure 3 on page 3). See “Input EQ and dynamics” on page 4.

Output source

Choose the *Source* for the audio output. The source can be a mix bus (which can have multiple sources itself) or the MicroBook II’s signal generator.

Output volume controls

The *Output volume controls* (Figure 7) provides independent volume control for all of the MicroBook II’s outputs.

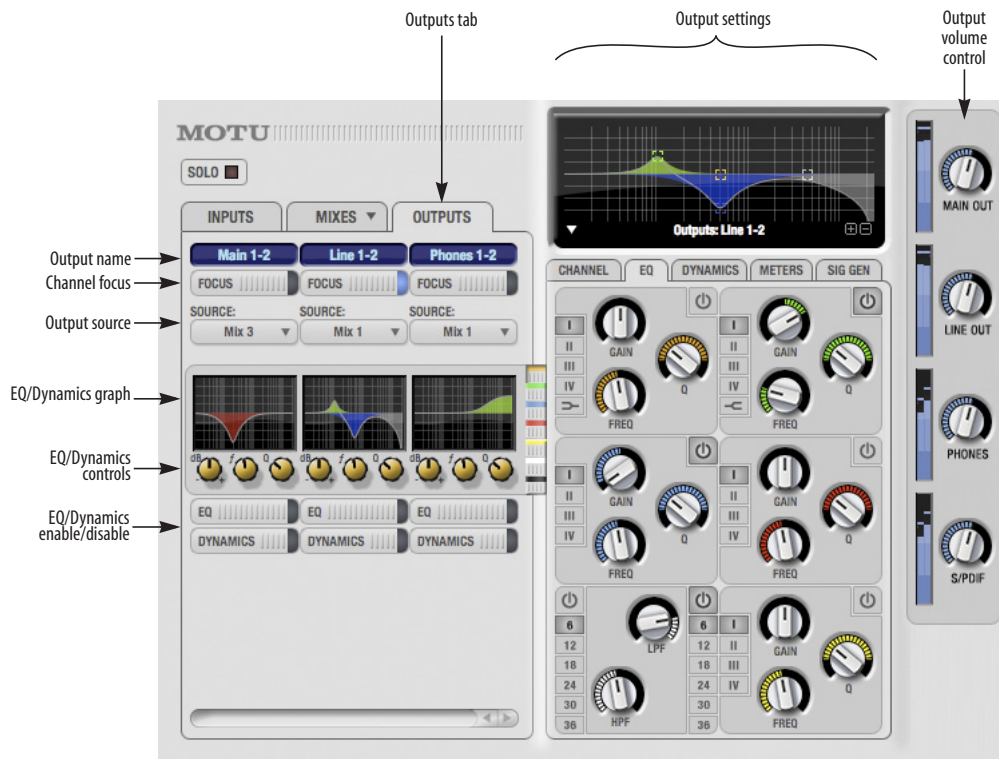


Figure 7: The Outputs tab.

CHANNEL SETTINGS SECTION

The channel settings section in the CueMix FX window (Figure 8) 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 Signal Generator, as shown below.

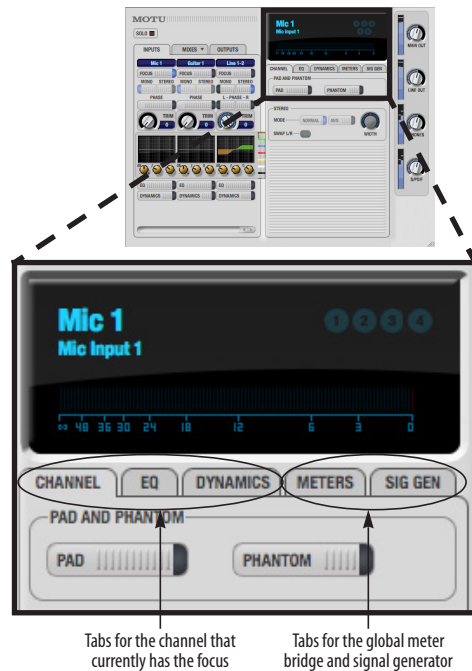


Figure 8: The Channel Settings section.

The Channel tab

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

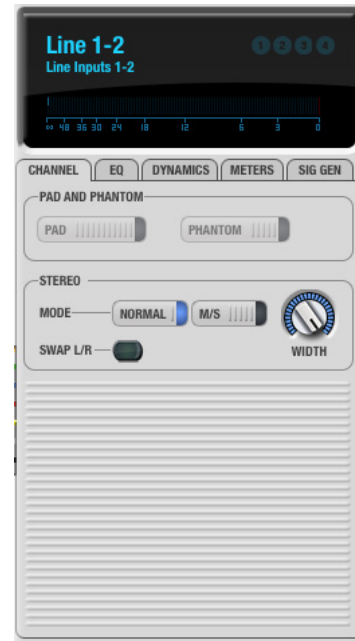


Figure 9: The Channel tab.

Signal flow

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

Pad and phantom

The Pad and phantom settings become active (ungrayed) when the focus is on the mic input.

Stereo settings

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

The *Width* knob (Figure 9) provides control over the stereo imaging, going from a full stereo image to mono (both channels panned equally). See “Width” on page 5.

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

Input meter and bus activity LEDs

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

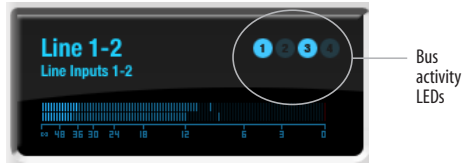


Figure 10: Input meter and bus activity LEDs.

The *input level meter* (Figure 10) is the same as the input meters in the Meters tab (Figure 13 on page 15) 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 MicroBook II. 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) show you which mix busses the input signal is being fed to. For example, LED #4 will glow under the following conditions: the input is unmuted in mix bus 4, its fader is up, and there is signal activity from the input going into the mix bus.

The EQ tab

The EQ tab (Figure 11) 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 MicroBook II *Vintage EQ section* (Figure 11) 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. The MicroBook II 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 3 on page 3 and Figure 7 on page 7). 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 11), allowing you to enable as few or as many bands as needed for each individual channel.

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

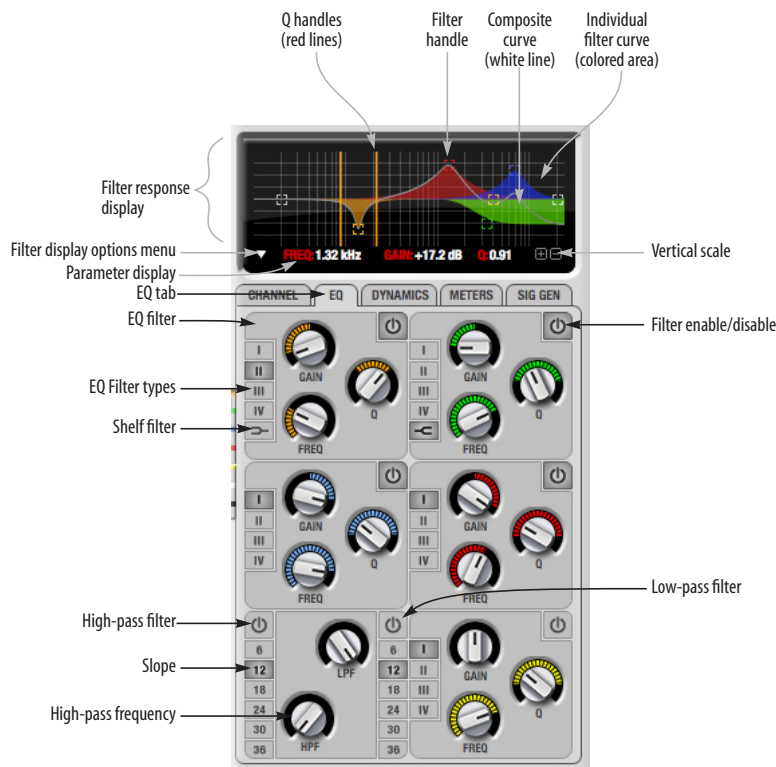


Figure 11: The EQ tab.

Vintage EQ Quick reference

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

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

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

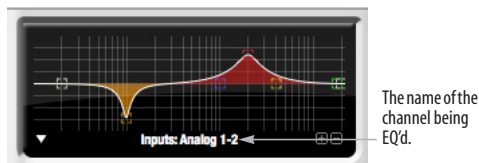


Figure 11-1: When a filter handle is not selected and when the cursor is not hovering over the display, the parameter display shows the name of the current channel being edited (the channel that currently has the focus).

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 11). 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 11).

Adjusting filters in the display

Each filter has a handle, displayed as shown below in Figure 11-2 (in the filter's color), for adjusting its boost/cut and/or frequency:

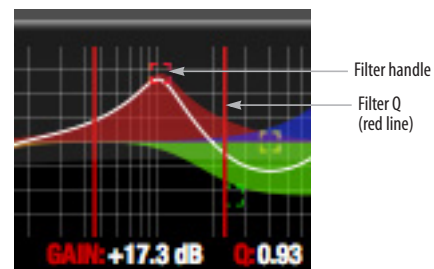


Figure 11-2: Drag the filter handle to adjust its frequency and/or boost/cut. Drag the Filter Q handles to adjust the Q.

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

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

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 11-3 through Figure 11-6), 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

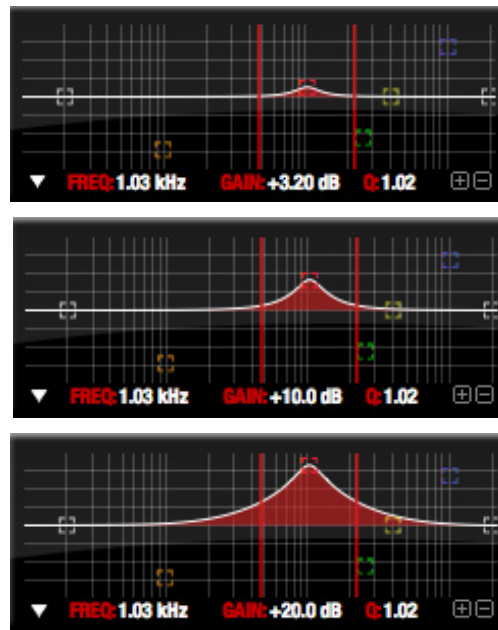


Figure 11-3: Type I EQ filter style.

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

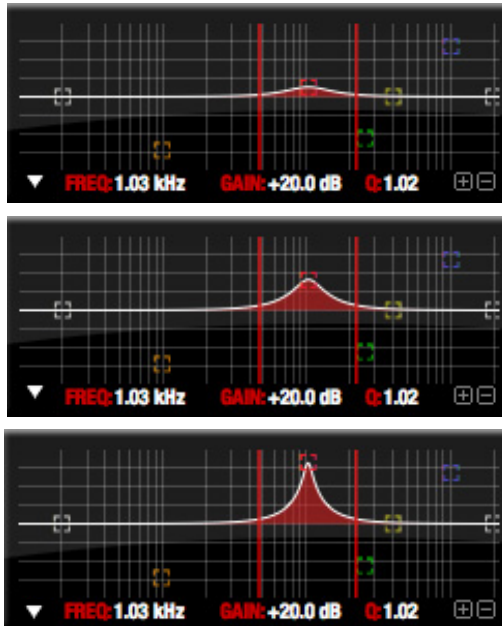


Figure 11-4: Type II EQ filter style.

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

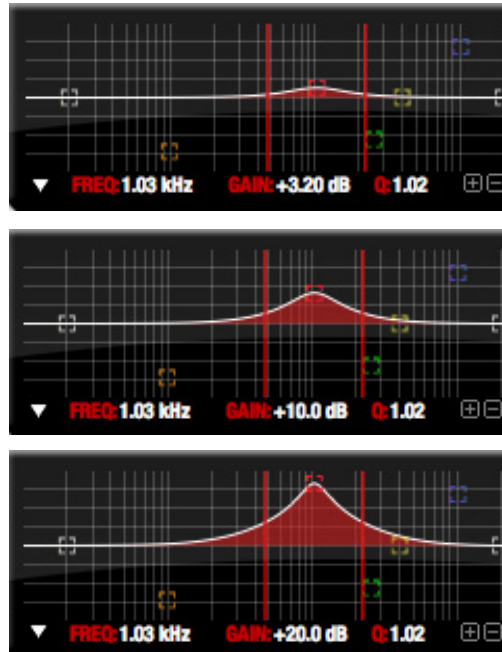


Figure 11-5: Type III EQ filter style.

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

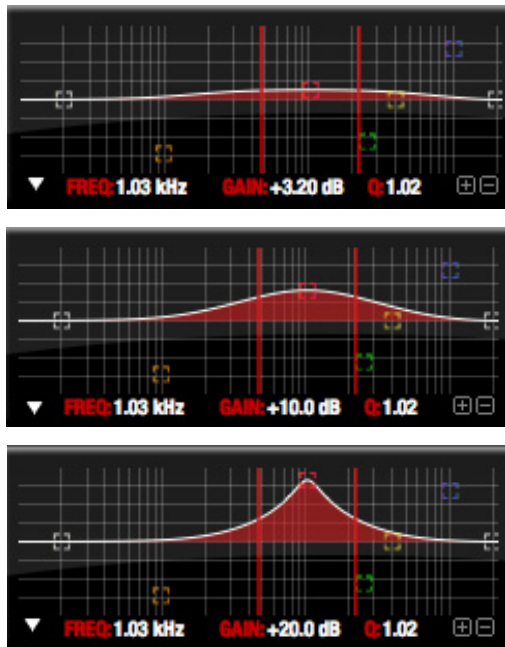


Figure 11-6: Type IV EQ filter style.

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

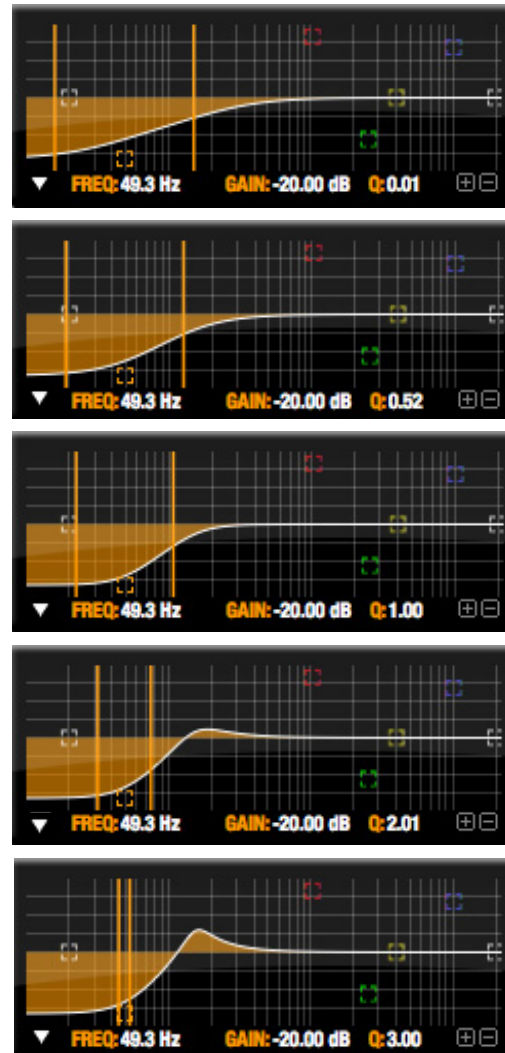


Figure 11-7: Shelf filter *Q* parameter overshoot.

When two top-most bands in the EQ tab are set to their *shelf filter* setting (Figure 11), the *Q* parameter controls the amount of overshoot applied to the response curve, as illustrated in Figure 11-7. 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 11-7 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:

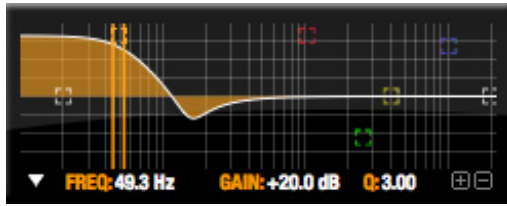


Figure 11-8: Overshoot when low shelf boost is applied.

Overshoot is also applied to high shelf boost and cut:



Figure 11-9: 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.

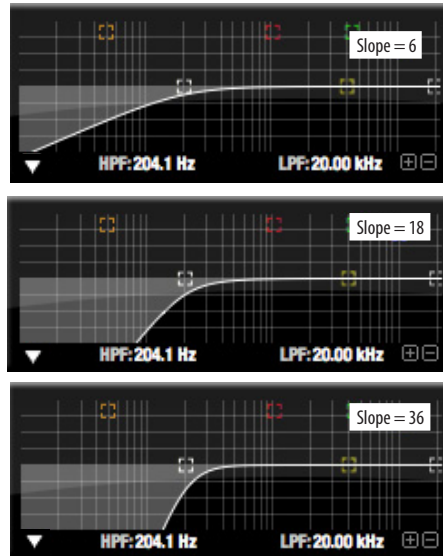


Figure 11-10: The low pass filter with three example slope settings.

The Dynamics tab

The Dynamics tab (Figure 12) 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.

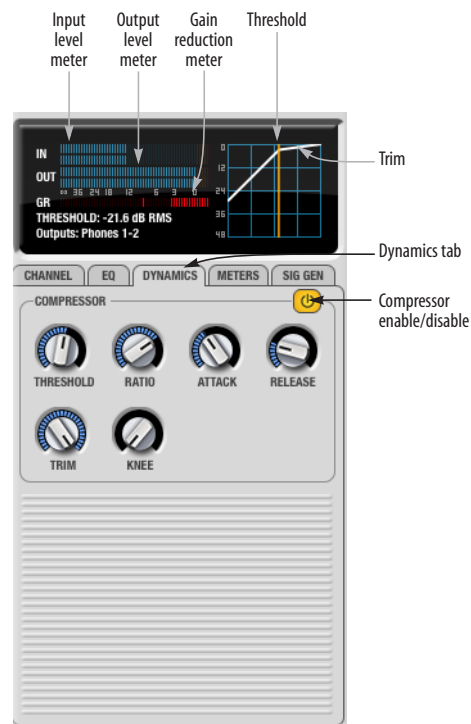


Figure 12: The Dynamics tab.

Enabling Dynamics

Each input and output channel has a global *Dynamics enable/disable* button (Figure 3 on page 3). This button enables or disables all dynamics processing for the channel.

Compressor

The *Compressor* (Figure 12 on page 14) 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.

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 12 on page 14).

Input level meter

The *Input Level* meter (Figure 12 on page 14) shows the level of the input signal before it enters the compressor.

Gain reduction (GR) meter

The Gain reduction (*GR*) level meter (Figure 12 on page 14) displays the current amount of attenuation applied by the compressor.

Output level

The *Output Level* meter (Figure 12 on page 14) displays the peaks of the output signal. Trim is applied before the Output Level meter.

METERS TAB

The *Meters* tab (Figure 13) serves as a comprehensive meter bridge for all inputs, outputs and mix busses in the MicroBook II, giving you a “bird’s-eye” view of all signal activity. It is ideal for confirming your signal routing programming and for troubleshooting.

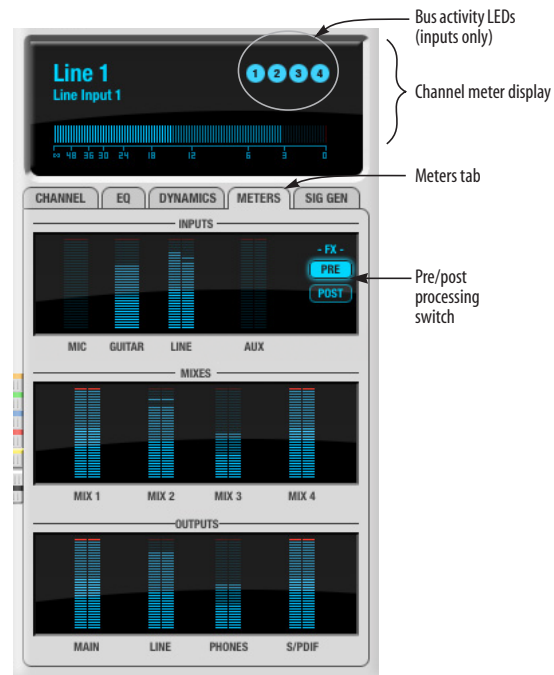


Figure 13: The Meters tab.

Channel meter display

The *channel meter display* (Figure 13) 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 13) are present only for inputs. See Figure 10 on page 9.

Pre/post processing switch

The *pre/post processing* switch (Figure 13) 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.)

SIGNAL GENERATOR TAB

The *Signal Generator* tab (Figure 14) provides settings for the Sine Wave generator (frequency and amplitude) and the White/Pink noise generators (amplitude). Drag a knob to adjust its setting; hold the Option/Alt key while dragging for fine control.



Figure 14: The Signal Generator tab.

Use the Output Source menus in the Outputs tab (Figure 7 on page 7) to choose the desired outputs for these test signals.

OUTPUT VOLUME CONTROLS

The Output Volume Controls provide final gain-stage control of the MicroBook II's outputs.



Figure 15: Output volume controls.

Individual output level knobs

Each of the MicroBook II's output pairs has an output level knob that can be adjusted independently.

DEVICE SETUP

Choose *Devices menu > MicroBook II > Setup* to access the *Scope/Return Channel Selection* settings below (Figure 16). The MicroBook II sends two extra return channels to the host audio software running on your computer. This window lets you control the audio source for each channel and whether or not the signal will be processed with the MicroBook EQ and/or Dynamics processing settings via the *Pre/post effects* menu.

Note that you can route entire mix busses, and even MicroBook II output signals, back to the computer. Just choose the desired mix bus or output from the *Source* menus.

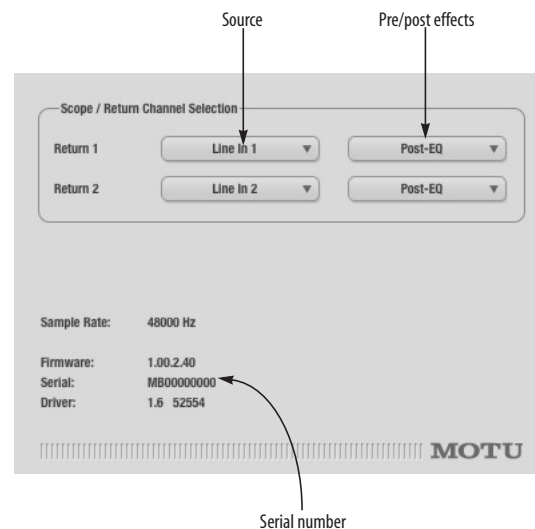


Figure 16: MicroBook II Device Setup.

Return channels

The source for the *Return 1* and *Return 2* channels in your host audio software (provided by the MicroBook Core Audio and ASIO driver) are determined by Scope/Return channel settings (Figure 16).

For example, if you want to mix inputs with effects and send the result to MicroBook II outputs, but you also want to send those same inputs dry (no processing) to the host, then select those inputs in the Scope/Return menus and choose *Return 1* and/or *Return 2* as the input to your host software.

Scope channels

The CueMix FX tuner and audio analysis (scoping) features described on the next few pages always monitor audio from the two Scope/Return channels (Figure 16).

Device information

The bottom portion of the Setup window displays the firmware version and serial number of the MicroBook II, as well as the installed version of the MicroBook II driver.

Host Buffer Multiplier (Windows only)

The *Host Buffer Multiplier* setting (Windows only) affects how the MicroBook II communicates with your host application software. The default value of “2 (4 ms)” will be fine under most circumstances. Only if you experience drop-outs or other audio problems should you choose a larger value, as raising this setting will increase software monitoring latency.

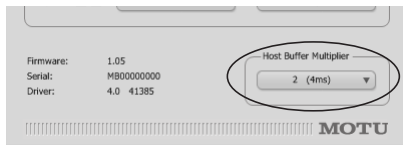


Figure 17: Host Buffer Multiplier

FFT AND SPECTROGRAM DISPLAY

FFT and spectrogram information can be displayed in the Filter response display section or as a separate window.

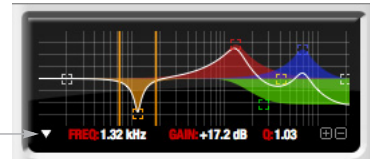


Figure 18: Filter Display options menu

Filter display options

The *Filter display options* menu (Figure 18) 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.

To show the FFT or spectrogram display in this window, the focused channels must be a stereo pair, and they must match the scope input channels, selected in the Device Setup dialog.

FFT display

Choose *Show FFT* from the *Filter display options* menu (Figure 18) to superimpose a real-time Fast Fourier Transform (FFT) frequency measurement curve over the EQ filter display, as demonstrated in Figure 19:

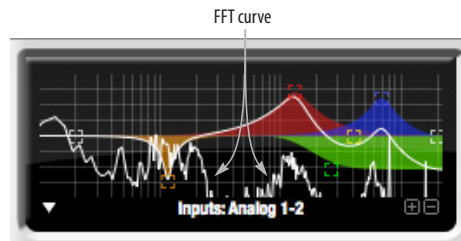


Figure 19: FFT display.

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 to toggle between the EQ'd and non-EQ'd FFT display.

Spectrogram

Choose *Show Spectrogram* from the *Filter display options* menu (Figure 18) to superimpose a real-time spectrogram “waterfall” display in the background of the EQ filter display, as demonstrated in Figure 20:

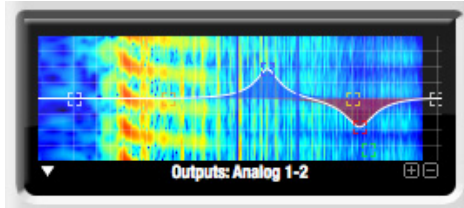


Figure 20: Spectrogram display.

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 21: Spectrogram color-to-amplitude spectrum.

Opening the FFT Analysis window

Choose *FFT Analysis* from the *Devices* menu (Figure 23) to open a new window with the filter EQ display for detailed inspection and adjustment of the EQ filter, as shown in Figure 22 on page 18.

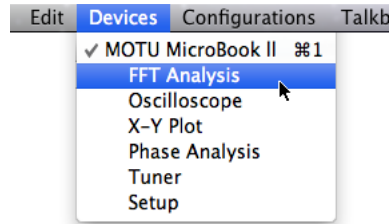


Figure 23: Opening the FFT Analysis window

View controls

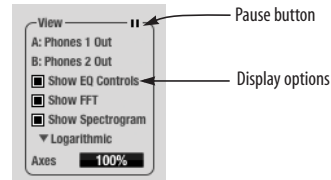


Figure 24: View controls

You can show and hide the FFT display, spectrogram or EQ band response curves as desired using the *View* controls (Figure 24). These settings are independent of the small graph display options (Figure 18), so you have the flexibility to display different combinations in each graph.

☛ “Enable EQ Controls” will be available only if the selected scope inputs are a stereo input pair or stereo output pair.

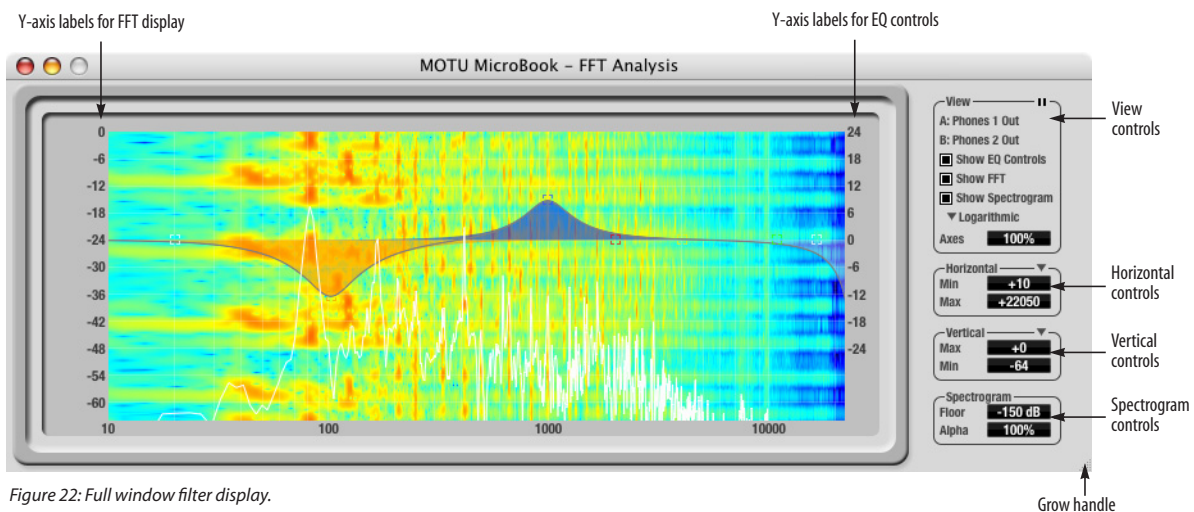


Figure 22: Full window filter display.

Logarithmic or Linear X-Axis Scale

The x-axis defaults to a logarithmic scale, but can be changed to a linear scale if desired. In the View controls (Figure 24), click *Logarithmic* to access the x-axis scale options menu. With a linear scale selected, frequency is constant, but the width of each octave along the x-axis is different. With a logarithmic scale selected, octaves are displayed with a constant width, but frequency is displayed logarithmically within each octave.

Axes display

The *Axes* control (Figure 24) sets the opacity of the grid displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

Pausing the display

The *Pause* button in the upper right corner of the View section (Figure 24) allows you to freeze the display at any time. To resume, click the button again.

The info box

When any EQ filter point is selected or dragged in the full window graph, the info box is shown next to the point in the full graph display (Figure 25).

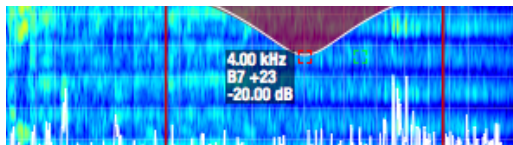


Figure 25: The Info Box.

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.

Horizontal controls (frequency axis)

The *Horizontal* controls (Figure 26) configure the value range of the x-axis (frequency). Click and drag the values up or down to set them, or double-click to return to the default value.

There are two modes for the controls: *Zoom/Offset* and *Min/Max*. To change the mode, use the Horizontal control menu (Figure 26).

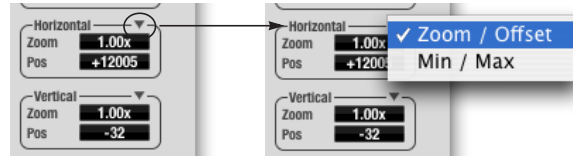


Figure 26: Horizontal control menu

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

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

Vertical controls (amplitude axis)

The *Vertical* controls (Figure 22) operate similarly to the Horizontal controls, except that they configure the y-axis (amplitude).

In *Zoom/Offset* mode, *Zoom* sets the display zoom from 1x to 100x, and *Pos* sets the center amplitude of the graph.

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

Spectrogram controls

The *Floor* control (Figure 22) sets the amplitude threshold for the spectrogram display, from -144 dB up to 0 dB.

The *Alpha* control (Figure 22) sets the opacity of the spectrogram information displayed in the graph, from 100% (fully visible) down to 0% (fully hidden).

OSCILLOSCOPE

The Oscilloscope (Figure 27) 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 27, 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 MicroBook II has its own oscilloscope. To open an oscilloscope, choose the *Oscilloscope* item from the Devices menu under the desired interface.

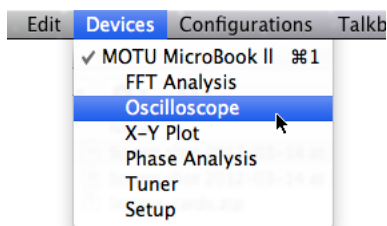


Figure 28: Opening the oscilloscope

Choosing a channel to display

The oscilloscope follows the *Scope/Return* channels selected in the Device Setup window. For more information, see “Scope channels” on page 16.

View controls

The View controls (Figure 29) provide several options for the oscilloscope display.

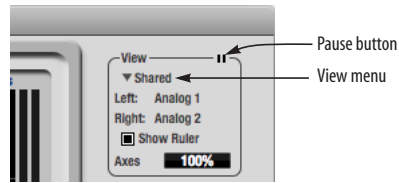


Figure 29: View controls

View menu

The View menu (Figure 29) 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 29) 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 22).

Pausing the display

The Pause button in the upper right corner of the View section (Figure 29) 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.



Figure 27: Oscilloscope

Horizontal controls (time axis)

The *Horizontal* controls (Figure 30) 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 30).

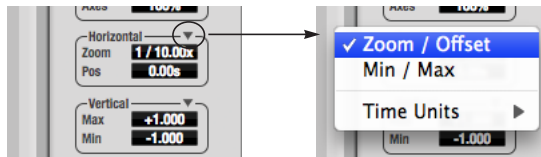


Figure 30: Horizontal control menu

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 30) provides the option to view the X axis in Seconds or Samples.

Vertical controls (amplitude axis)

The *Vertical* controls (Figure 30) 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 200x, 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.

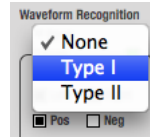


Figure 31: Waveform Recognition menu

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 32) 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.

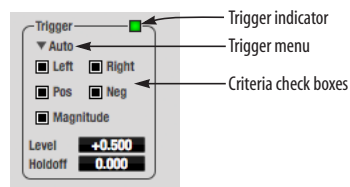


Figure 32: Trigger settings

Criteria

The criteria checkboxes (Figure 32) determine the conditions that the trigger is looking for and where it will look for them.

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

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

event where amplitude is decreasing. One or both of these can be enabled simultaneously. If neither is enabled, the criteria will not be found because the trigger is not looking for any particular kind of event.

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

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

Holdoff

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

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

Trigger modes

The Trigger menu (Figure 32) 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 32 on page 21) or by pressing the spacebar.

Trigger indicator

The Trigger indicator (Figure 32) displays the state of the trigger, and also provides a way to manually interact with it. The Trigger indicator always displays one of three colors:

Color	Status
Green	When the current Trigger criteria has been met (including when the Trigger mode is <i>None</i>).
Yellow	When the Trigger is armed, but has not yet found an event which matches its criteria. Yellow can also indicate that the graph has been manually paused using the Pause button in the View section (see “Pausing the display” on page 20).
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.

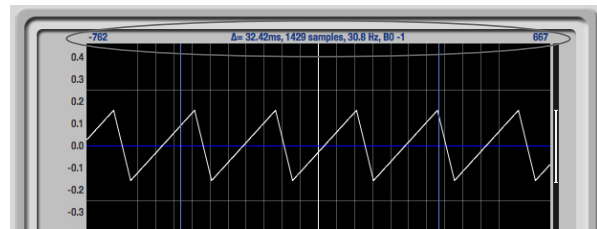


Figure 33: Measurement information

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 34. In this example, a snare hit is being compressed by Digital Performer’s Dynamics plug-in. As you make adjustments to the compression plug-in’s settings, you can see the transient waveform change the next time the Oscilloscope triggers. For compression, this can be particularly useful for balancing the effect of the attack on the transient, relative to the decay portion of the waveform. Conversely, you can see the effect of the threshold setting directly on the decay portion, relative to the attack. In effect, you can see as well as hear the results of your compression adjustments.

To view a transient waveform in the Oscilloscope display, turn off Waveform Recognition and use the *Normal* Trigger mode. Adjust the level high enough to encompass the vertical amplitude of most of the transient. If the transient pulse sweeps across the screen, try raising the Holdoff level. Once

the transient is settled in the display and fairly stable, you may need to adjust the horizontal position to center it in the display. These settings are depicted in the example in Figure 34.

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 32 on page 21), choose *Single Sweep* from the trigger menu (Figure 32), set the level to 0.999 and click the trigger indicator (Figure 32) 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*.

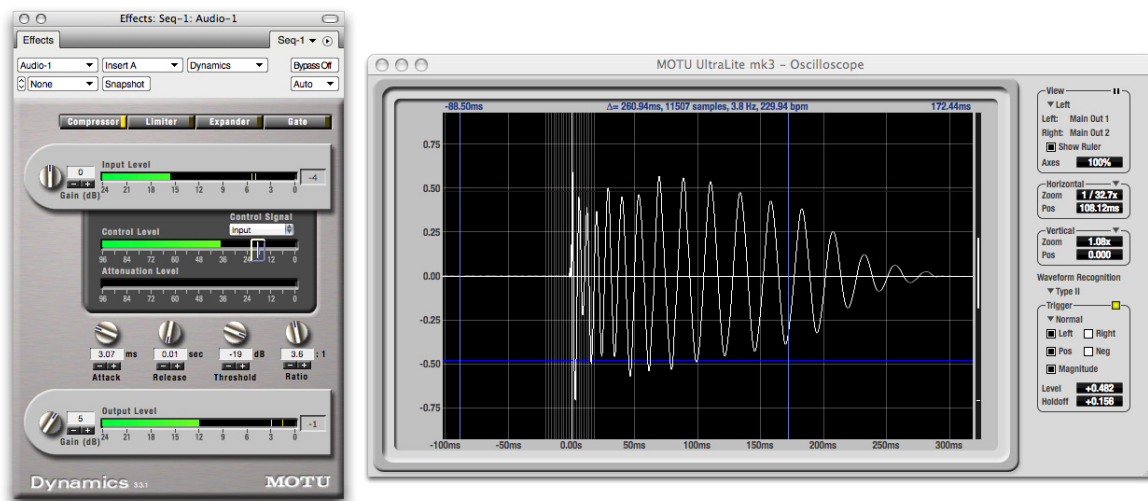


Figure 34: Viewing transients in the Oscilloscope

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

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

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

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

Monitoring control voltage output from Volta

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

For more information on Volta, see www.motu.com.

X-Y PLOT

The *X-Y Plot* window (Figure 35) 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 35, below). There are also thick white diagonal lines for $y = x$ and $y = -x$.

Metering

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

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

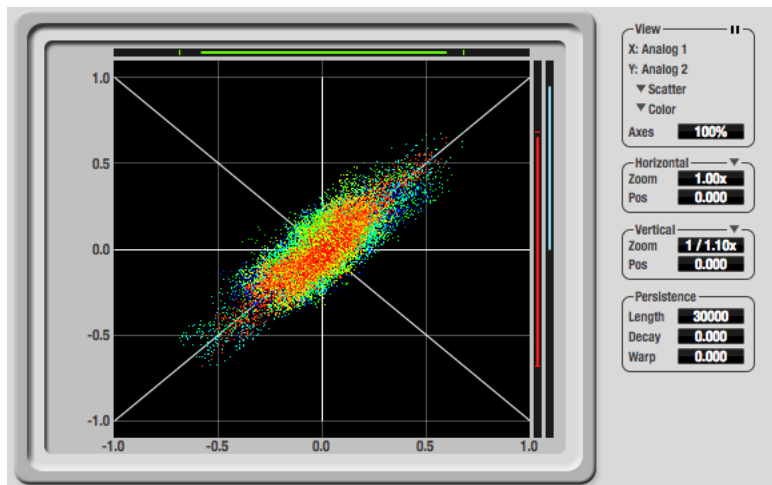


Figure 35: X-Y Plot

Opening the X-Y Plot

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

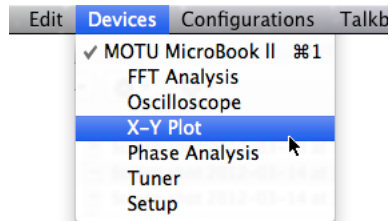


Figure 36: Opening the X-Y Plot.

Choosing a channel pair to display

The X-Y Plot follows the *Scope/Return* channels selected in the Device Setup window. For more information, see “Scope channels” on page 16.

View controls

The View controls (Figure 37) provide several options for the X-Y Plot display.

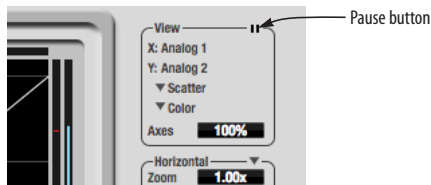


Figure 37: View controls

Pausing the display

The Pause button in the upper right corner of the View section (Figure 37) allows you to freeze the display at any time. To resume, click the button again. The level meters will remain active while the display is paused.

Line/Scatter

Choose either *Line* or *Scatter* from the menu in the View section (Figure 37) 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 38.

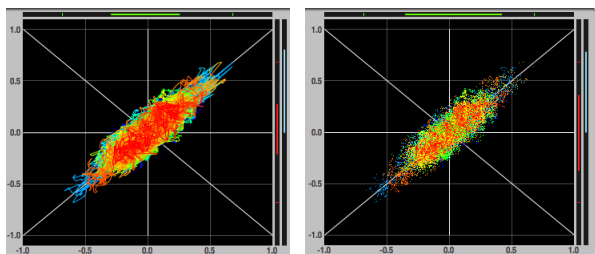


Figure 38: The same X-Y Plot displayed in Line versus Scatter mode.

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 37) 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 26.

Axes

The *Axes* control (Figure 37) 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 39) 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 39.

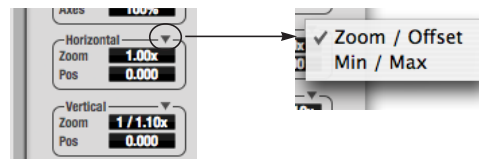


Figure 39: Setting the Horizontal or Vertical control modes.

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 40) affect the appearance of data from when it is first displayed until it disappears from the grid.

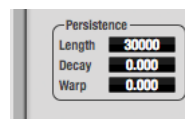


Figure 40: The Persistence controls.

Length

Length (Figure 40) 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 40) 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 40) 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 $y = x$ 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.

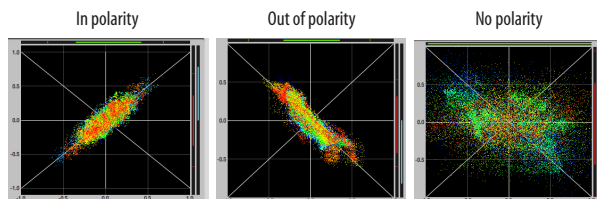


Figure 41: Checking polarity in a stereo signal with the X-Y Plot.

PHASE ANALYSIS

The *Phase Analysis* window (Figure 44 on page 27) 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 MicroBook II has its own Phase Analysis window. Choose the *Phase Analysis* item from the Devices menu under the desired interface.

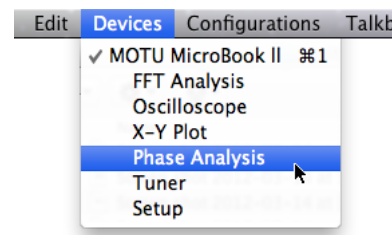


Figure 42: Opening the Phase Analysis.

Choosing a channel pair to display

The Phase Analysis window follows the *Scope/Return* channels selected in the Device Setup window. For more information, see “Scope channels” on page 16.

View controls

The View controls (Figure 43) provide several options for the Phase Analysis display.

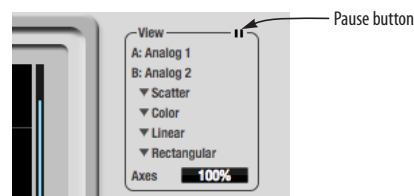


Figure 43: View controls

Pausing the display

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

A/B (stereo audio channels)

The View section (Figure 43) 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 43) 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 45.

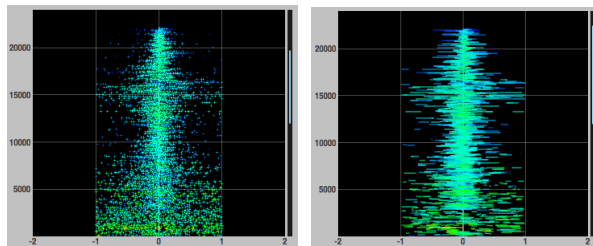


Figure 45: The same Phase Analysis displayed in Line versus Scatter mode.

☛ 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 28).

Color/Grayscale

In *Color* mode (Figure 43) 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 43) 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 43 on page 26) 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.

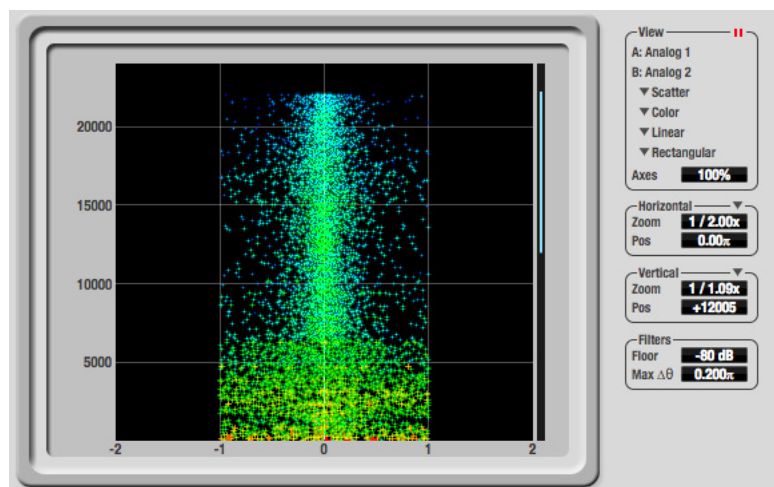


Figure 44: Phase Analysis

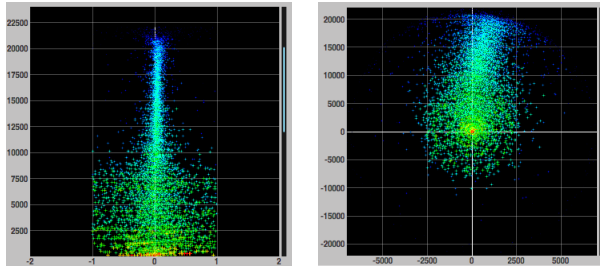


Figure 46: Rectangular versus Polar display (with a linear plot).

Above, Figure 46 shows Rectangular versus Polar display with a Linear plot. Below, Figure 47 shows the same displays (and the same data) with a Logarithmic plot:

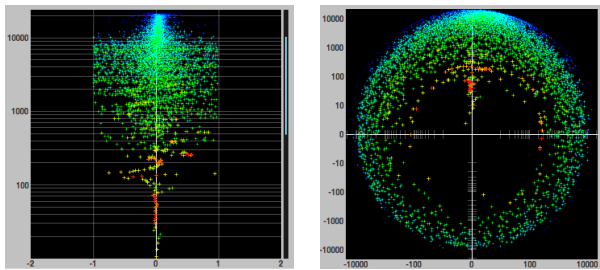


Figure 47: Rectangular versus Polar display with a logarithmic plot.

Axes

The *Axes* control (Figure 43 on page 26) 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 48) 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 48.

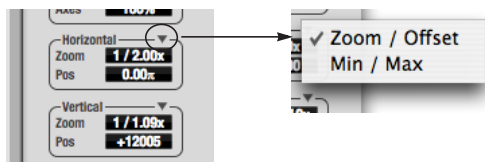


Figure 48: Setting the Horizontal or Vertical control modes.

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 49) lets you control the density of the Phase Analysis display.

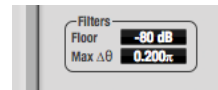


Figure 49: Filters

Floor

Floor (Figure 49) 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 49) only affects Line view (see “Line/Scatter” on page 27) and sets the maximum difference in phase 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 50 on page 29), stereo material that is predominantly phase-aligned (correlated) appears along the vertical axis, as demonstrated in the first column (*Perfectly in phase*) in Figure 50. 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 50. Delays appear as spirals in the polar display.

The rectangular display (bottom row of Figure 50) 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 50.

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 51 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).

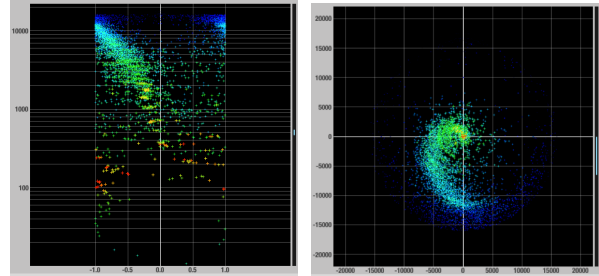


Figure 51: A stereo mix with phase issues.

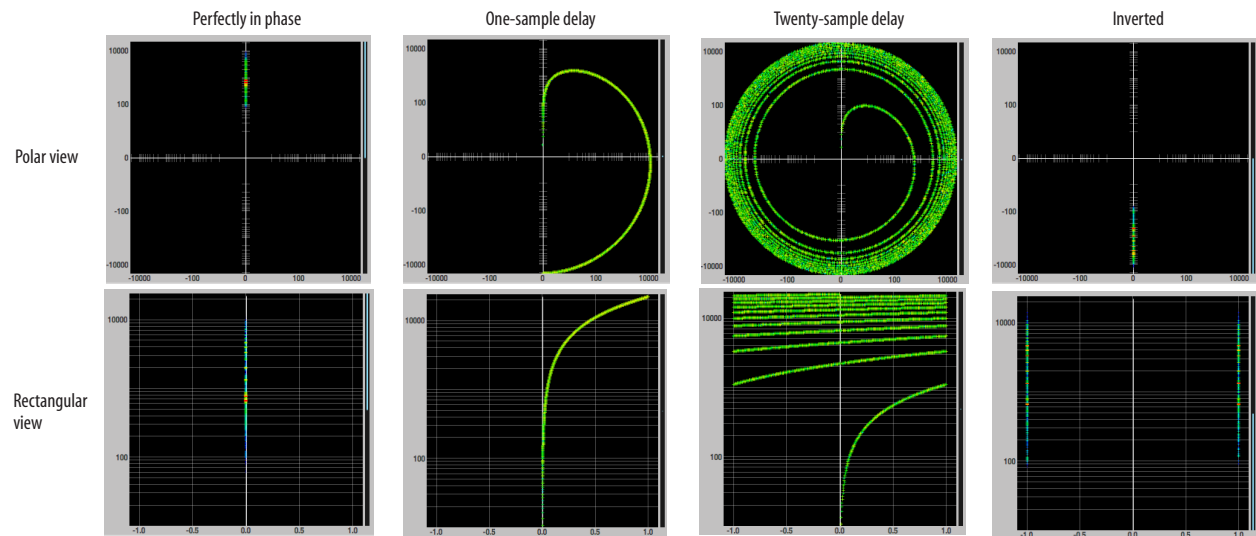
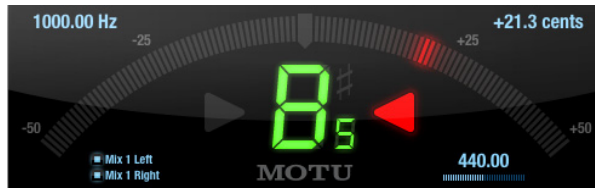


Figure 50: Two identical audio streams in the Phase Analysis.

TUNER



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

Opening the Tuner

Each MicroBook II has its own *Tuner* window. Choose the *Tuner* item from the *Devices* menu under the desired interface.

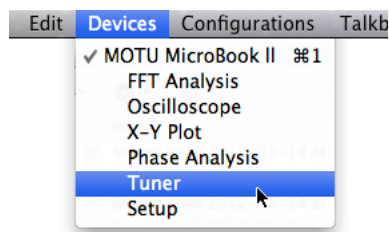


Figure 52: Opening the *Tuner* window.

Choosing a channel pair to display

The *Tuner* window follows the *Scope/Return* channels selected in the *Device Setup* window. For more information, see “*Scope channels*” on page 16.

Choosing the channels to tune

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

Tuner controls

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

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

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

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

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

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

Tuning stereo signals

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

SHORTCUTS

Hold down the following modifier keys as shortcuts:

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

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

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

FILE MENU

Save and Load Hardware Preset

The *Save Hardware Preset* and *Load Hardware Preset* items apply to other MOTU interfaces and have no effect on the MicroBook II.

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 Output

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

Hardware follows console stereo settings

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

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 6 on page 5) and choose Copy from the Edit menu, or press Command-C (Mac) or Control-C (Windows).
- 2 Choose the destination mix and choose Paste from the Edit menu, or press Command-V (Mac) or Control-V (Windows).

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 the signal analysis tools: FFT Analysis, Oscilloscope, X-Y Plot, Phase Analysis, and Tuner. Choose one to open its window. For details on these features, see page 17.

Additionally, a Setup command is provided to access additional device-specific settings for each MicroBook II. See “Device Setup” on page 16.

CONFIGURATIONS MENU

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

Here is a summary of Configurations menu operations:

Configurations Menu item	What it does
Create New	Lets you name and save a new configuration, which appears at the bottom of the Configurations menu.
Save	Overwrites the current configuration (checked in the list at the bottom of the menu) with the current settings in CueMix FX.
Save To	Same as Save above, except that it lets you first choose the configuration you wish to save to (instead of the current one).
Delete	Lets you choose a configuration to permanently remove from the menu.
Import	Loads all configurations from a configuration file on disk.
Export	Saves all current configurations as a file on disk.
Factory Defaults	Resets all hardware settings to their factory defaults, except Sample Rate (Mac and Windows) and Host Buffer Multiplier (Windows only).
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.

PHONES MENU

The Phones menu applies to other MOTU interfaces and does not pertain to the MicroBook II.

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.

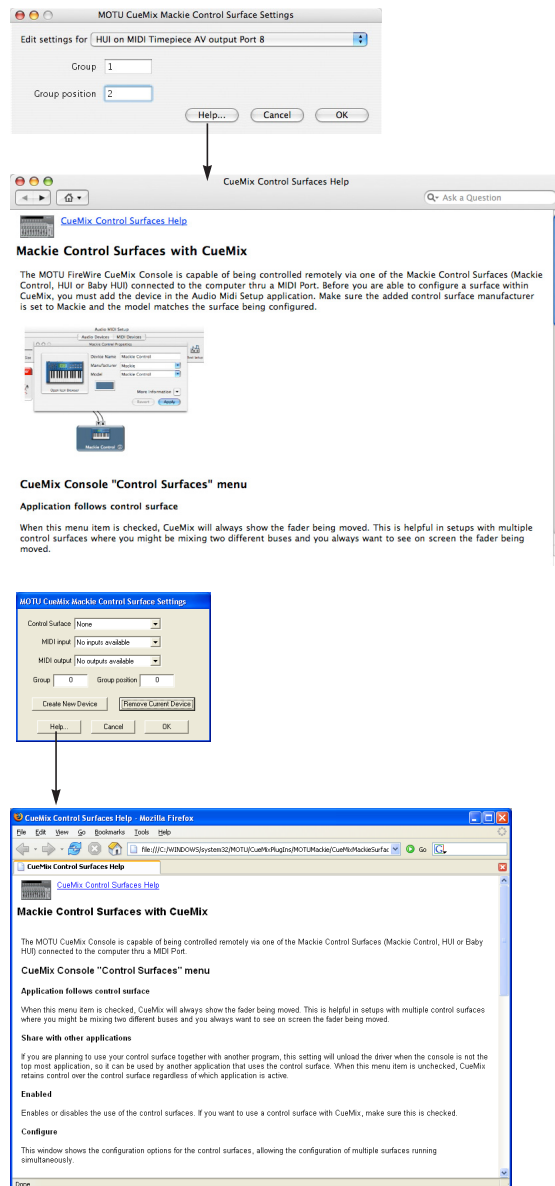


Figure 53: Refer to the extensive on-line help for details about configuring CueMix FX for operation with your control surface product.

Other HUI-compatible control surfaces

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

Other control surface hardware products

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