User Manual

October 20, 2025

WING, WING COMPACT, WING RACK

Firmware 3.1

behringer

Do not read this

This document is continually maintained and improved. If you have any comments or questions, you are welcome to share them at ideas.behringer.com.

Keep in mind that the console's GUI may differ from the screenshots in this guide as new firmware versions are released.

Contents

Section 1

Introduction

Congratulations on purchasing the groundbreaking WING console! This manual provides a detailed description of WING's funtionality and effects. Additional guides and videos are available on behringer.com and through our social media channels.

We recommend checking behringer.com regularly for firmware updates, as new features and bug fixes are released frequently. For details on the update process, please refer to Chapter 9 of this guide. Our development team is eager to hear your feedback at ideas.behringer.com, where you can share suggestions and discover upcoming improvements and features.

Please note that the console's GUI may differ from the screenshots in this guide as new firmware versions are released.

Section 2

Sources, Channels, Buses and Outputs

With its enhanced routing flexibility, WING introduces a new method for labeling signal Sources using names, icons, and colors, alongside their physical parameters such as gain, phantom power, and phase inversion.

WING Sources can be routed to one or more channels for signal processing, or routed to Main Buses, Buses, and Matrices. They can also be sent directly from any input to any output directly, when no processing is needed, such as in recording setups or when sharing audio with another console for independent mixes. Additionally, Sources enable remote control of preamps and allow for the sharing of names and customization settings via AES50.

WING Sources include the following information:

- Identifying characteristics like name, color, icon, and tags
- Physical characteristics such as gain, mute, phase inversion and phantom power
- Mono, stereo or mid/side configuration. All 48 channels on the console can process mono, stereo and mid/side signals, without the need to link two adjacent channels. See Section 5.4: ROUTING for more information.

2.1 Sources

Inputs can be analog signals via combo jacks or 1/4-inch TRS connectors on the rear panel, or digital signals via StageConnect, USB, AES50, internal modules, and expansion cards.

WING has the following external SOURCE GROUPS:

- Local In (8 on-board preamps on WING, 24 on WING COMPACT and WING RACK)
- Aux In (8 TRS connectors for line-level inputs, WING only)
- AES/EBU (2 channels)
- AES50 A, B and C (48 channels each)
- StageConnect (up to 32 channels)
- USB Audio (48 channels)
- External expansion card (WING-LIVE by default) (64 channels)
- Internal module (64 channels)

- USB Player (4 channels)

2.2 40 Input Channels and 8 Aux Channels

WING channels offer powerful and flexible processing. All Input Channels (1-40) support mono, stereo, or mid/side signals without requiring adjacent channel linking. Aux Channels (A1-A8) support mono and stereo signals. As a result, all signal processors and sends within each channel are stereo capable.

When a mid/side Source is assigned to a channel, it is decoded into standard stereo before processing.

Each Input or Aux channel can accommodate both a MAIN and an ALT (alternative) Source. Channels automatically adopt the Source's customization properties and mono/stereo configuration (see Section 5.1 → ICON / COLOR for details).

2.3 4 Stereo Main Buses

WING features 4 stereo Main buses (M1-M4). Each channel has 4 independent Main sends that can be configured as Pre or Post fader (Fig. 1).

A typical application for multiple Main buses is using a single WING console to mix both the PA system in a venue and a live stream of the same event. One Main bus can handle the PA mix, while another handles the stream mix.

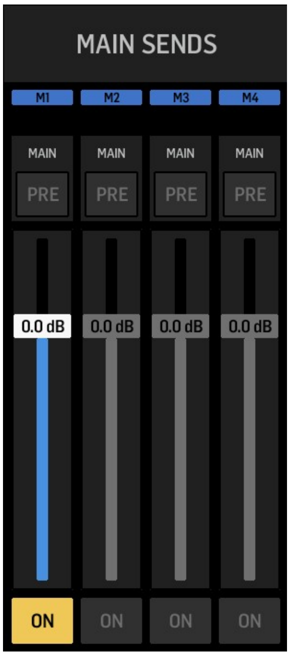

Figure 1: MAIN sends on a channel

2.4 16 Stereo Buses

WING provides 16 stereo buses (B1-B16) that can operate in three modes:

- TAP: The signal sent to the bus is derived from the channel's TAP point, which is freely customizable on Input Channels and fixed at the pre-fader position on other channel types (see Section 6.1 Input Channels for details). This mode is commonly used for monitoring mixes by setting the TAP point to a pre-fader position.

- POST: The signal sent to the bus is derived post-fader, typically used for adding effects like reverb or delay.
- GROUP: The send level to the bus is deactivated and controlled solely by the channel fader. This mode is often used for group processing, such as compressing all drum channels together.

All channels can be sent in different modes to the same bus. For example, one channel can be sent in Group mode to Bus 1, while another is sent in TAP mode to Bus 1. This configuration can be set in the Bus Sends section of each channel (Fig. 2) or on the Bus Feed Configuration screen (Fig. 27). For more information, see Section 5.1 → Home, Bus Feed Configuration.

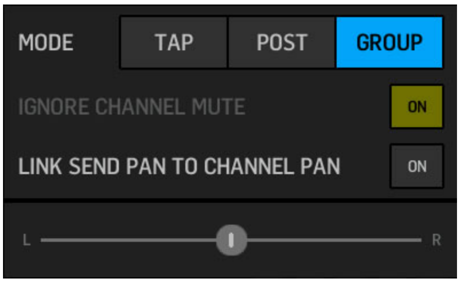

Figure 2: Channel Send settings

2.5 8 Matrix Buses

The 8 Matrix buses (MX1 - MX8) offer processing and routing options like the 16 Buses. However, Matrices can only be routed to digital or analog outputs, not to other Buses or Matrices. Additionally, Matrices can serve as Sources for any channel.

2.6 Internal Sources

Bus / Main / Matrix

All Buses, Mains and Matrices can be selected as the Source for all Input and Aux channels.

2 Oscillators

There are two independent mono test tone generators that can be configured for sine wave, pink noise or white noise output. They can be used as a Source for any channel or output.

24 User Signals

User Signals are copies of any of the 40 input channels, 8 Aux channels, 16 Buses, 8 Matrices or 4 Main buses. The signal can be derived pre-fader (from the TAP point on Input channels) or post-fader.

Each User Signal is mono. Users can choose whether the User Signal is a copy of the left (L) or right (R) channel, or a sum of both (L+R). If the original channel or bus is mono, this setting has no effect.

Two adjacent User Signals can be configured as a single stereo or mid/side Source, allowing for further processing and routing as needed.

32 User Patches

User Patches are copies of external Sources, either digital or analog, such as Local In, AES50, USB Audio, etc. User Patches allow you to create stereo or mid/side combinations of raw input signals that don't belong to the same Source Group or which are not adjacent. Possible applications are:

- Combining two signals from different Source Groups into a stereo or mid/side configuration
- Combining non-adjacent signals into a stereo or mid/side configuration
- Creating a stereo or mid/side Source that starts with an even channel
- Apply different gain or polarity to each channel of a stereo or mid/side configuration
- Apply phantom power to only one channel of a stereo or mid/side configuration

Each User Patch is mono. Two adjacent User Patches can be configured as a single stereo or mid/side Source and then processed or routed as desired.

Section 3

Rear Panel Connections

3.1 WING

The rear panel analog connections on WING (Fig. 3) include **8 Midas PRO microphone preamps**, **8 Midas PRO XLR outputs**, **8 balanced ¼" TRS AUX input** and **8 balanced ¼" TRS AUX output** connectors. **2 ¼" TRS stereo headphone** connectors are available below the top corners of the console, both providing a copy of the same signal.

A **lamp socket** accepts a standard 12 V light.

5-pin MIDI IN and **OUT** jacks allow external MIDI control, and a pair of **¼" TRS jacks** for up to **four GPIOs** allow basic input and output commands.

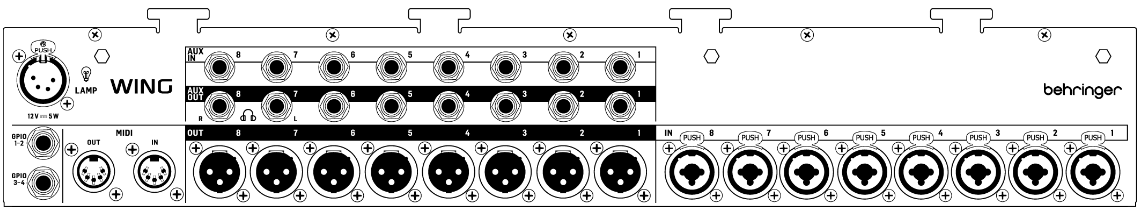

Figure 3: WING Analog I/O, MIDI and GPIO connectors

3.2 WING COMPACT

The rear panel analog connections on WING COMPACT (Fig. 4) include **24 Midas PRO microphone preamps** and **8 Midas PRO XLR outputs**. **2 ¼" TRS stereo headphone** connectors are available below the top and bottom right corners of the console, both providing a copy of the same signal.

A **lamp socket** on the top panel accepts a standard 12 V light. **5-pin MIDI IN** and **OUT jacks** allow external MIDI control, and a **¼" TRS jack** for **two GPIOs** allows basic input and output commands.

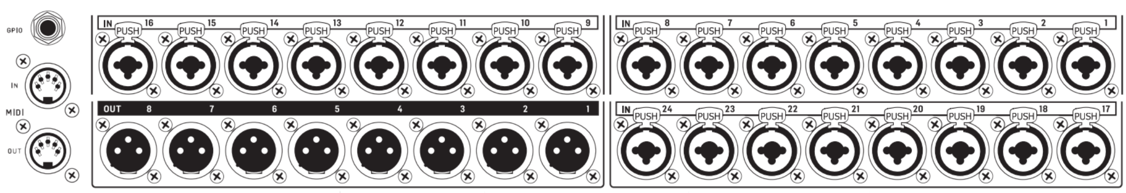

Figure 4: WING COMPACT Analog I/O, MIDI and GPIO connectors

3.3 WING RACK

The rear panel analog connections on WING RACK (Fig. 5) include **24 Midas PRO microphone preamps**, **8 Midas PRO XLR outputs** and **4 ¼" TRS connectors**, each with a high-performance **headphone amplifier**, providing hardwired copies of the signals routed to the XLR outputs. A fifth and independent headphone connector with its corresponding amplifier is available on the front panel.

5-pin MIDI IN and **OUT** jacks allow external MIDI control, and a pair of **¼" TRS jacks** for up to **four GPIOs** allow basic input and output commands.

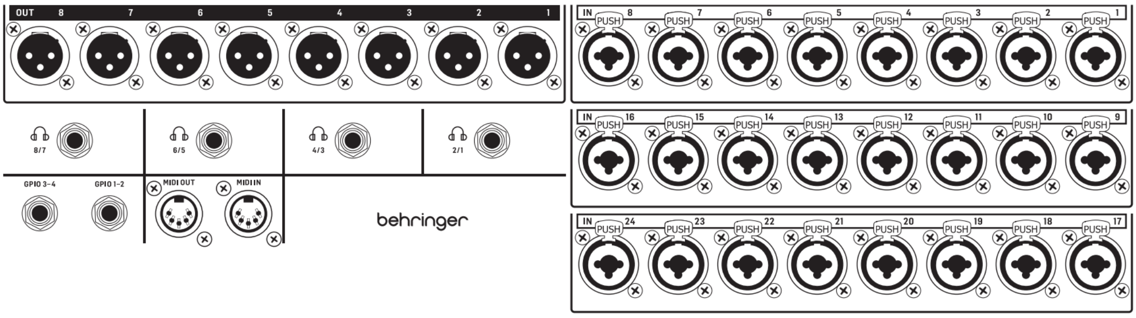

Figure 5: WING RACK Analog I/O, MIDI and GPIO connectors

3.4 Ethernet / AES50 / Control / StageConnect / AES/EBU

All WING models have the same digital I/O (Fig. 6).

2 Ethernet ports for remote control and Audio over IP (AoIP) communication from/to the internal module. Please note that internal modules are optional add-ons.

1 USB type B connector for the following uses:

- 48 input/48 output audio interface. The corresponding ASIO driver can be downloaded from behringer.com.
- MIDI DAW controller
- Firmware updates
- Data exchange (snapshots, snippets, channel presets, etc.)

3 AES50 ports, each with up to 48 input and output channels to and from digital stage boxes or other consoles, ensuring a high channel count and allowing patching to and from multiple locations. WING is compatible with Behringer X32 and Midas M32 mixers, as well as Behringer S series and Midas DL series stage boxes.

All AES50 connections between WING and stage boxes should meet the following requirements:

- Use shielded CAT-5e cables
- Feature rugged and lockable cable ends
- Have a maximum cable length of 80 meters

StageConnect transmits 32 digital audio channels (the number of inputs and outputs is user-definable) at 44.1/48 kHz and 24-bit resolution with sub-millisecond latency using a standard balanced XLR cable (110 Ohm impedance DMX-cable recommended) of up to 40 m in length. Use the StageConnect Calculator desktop application to plan and simulate configurations.

Stereo AES/EBU (AES3) input and output connections can be made via XLR cables.

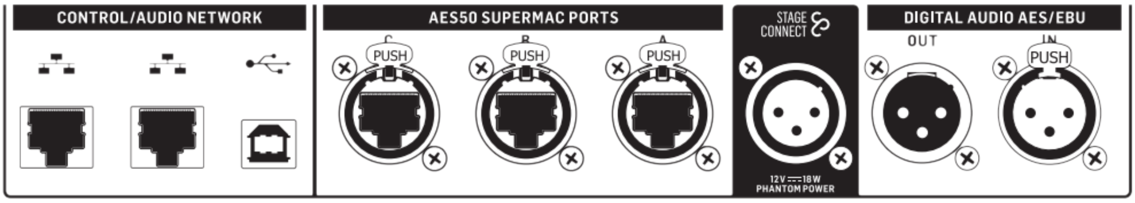

Figure 6: ETHERNET/AES50/Control/StageConnect connectors

3.5 Expansion Slot

All WING models include the WING-LIVE card (Fig. 7), which allows up to 64 channels of 48 kHz / 32-bit audio to be recorded onto a pair of SD or SDHC cards.

Expansion cards for other protocols such as Dante and MADI can be inserted in this slot. Check with your local retailer for available expansion card options.

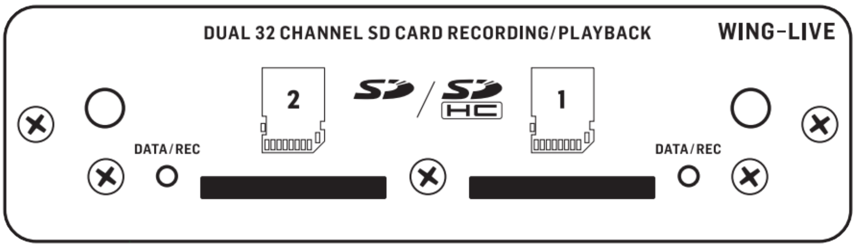

Figure 7: WING-LIVE expansion card

Section 4

Control Surface

All WING consoles have the same 10.1" touch screen that allows controlling all the console's parameters. Each model has different fader sections and control surfaces that are discussed in detail below.

4.1 Main Display

All WING controls can be edited via the 10.1" multi-touch screen main display (Fig. 8). Access to different screens is provided by the 7 buttons to the left of the display and the VIEW buttons located in each major section of the top panel (BK and COMPACT). An overview of each main screen is presented in Section 5 Main Screens.

The UTILITY button only lights up where it serves a specific function.

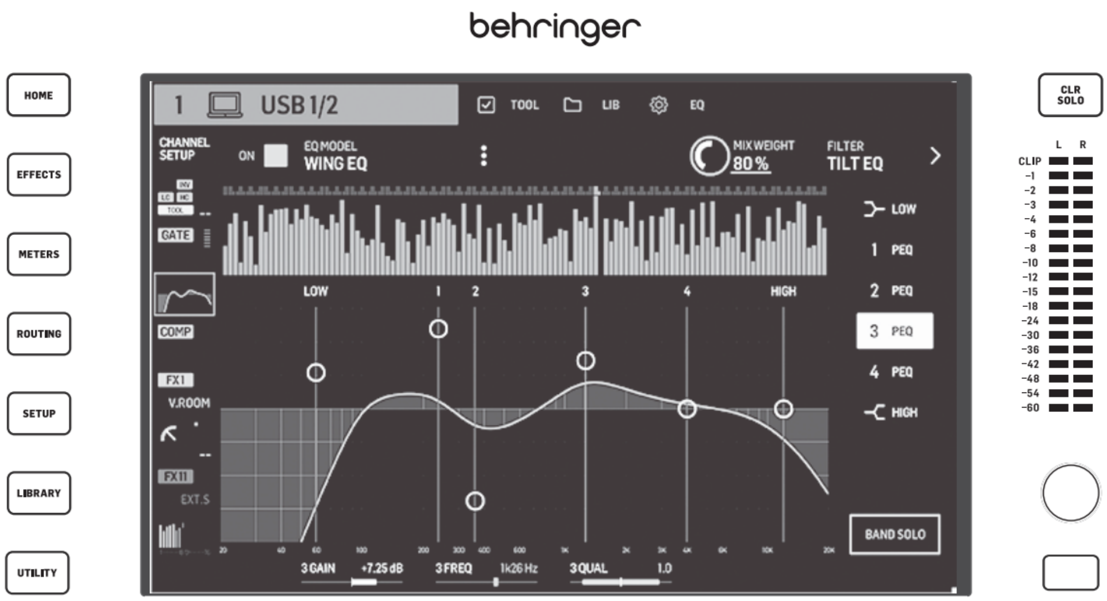

Figure 8: Main Display

Both WING and WING COMPACT feature 6 touch-sensitive knobs below the display for adjusting parameters shown at the bottom of the current screen. Touching a knob highlights the corresponding element on the screen. An additional context-sensitive knob to the right of the display on all models allows fine adjustments after selecting an item.

A multi-purpose, unlabeled button located below the additional knob performs various functions depending on the active screen, such as acting as a tap tempo button when editing delay effects.

The large stereo meter displays the levels of different buses, as configured in the Setup screen.

The CLR SOLO button releases all channels and buses currently active on the solo bus

4.2 Fader Sections and Layer Buttons

All WING models feature dedicated areas for specific fader layers. Selecting a different fader layer brings a new group of channels to the corresponding fader section.

Fader sections can display custom fader layers beyond those predefined on the Layer buttons. Each layer can be customized with any combination of channels, Aux channels, Buses, Mains, and Matrices; send levels to specific Buses, Mains, or Matrices; as well as individual effect parameters and MIDI CCs.

Most layers can include additional pages, which can be configured on the corresponding layer's CONFIG screen. Once set up, these pages are accessible via the shift arrows.

For more details, see Section 5.7 Overview and Section 5.10 CONFIG.

WING

WING has three fader sections (Fig. 9). The 12 faders on the left section are commonly used for input channels. The 8 faders in the center section typically control buses and DCAs, and the 4 faders on the right sections are commonly used for Mains and Matrices or to quickly adjust Input channel parameters on the 4-Channel Section (see Section 4.5).

If a layer exceeds the number of physical faders available in a section, the shift arrows will scroll in blocks of 4 channels (also 8 or 12, depending on the configuration in the Setup screen).

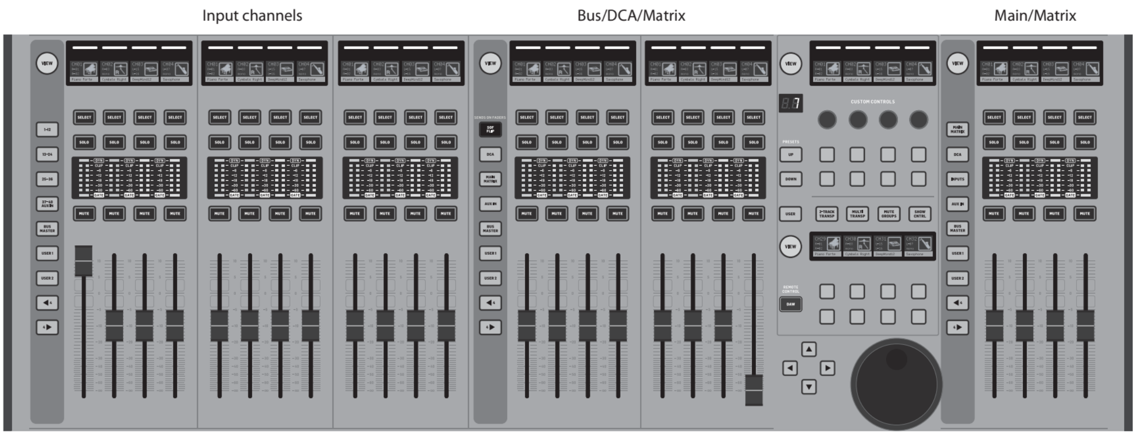

Figure 9: WING Fader sections

The layers are configured by default as follows:

Left fader section:

- 1-12: input channels 1 to 12
- 13-24: input channels 13-24
- 25-36: input channels 25-36

- 37-40 Aux In: input channels 37-40 and Aux channels 1-8
- BUS MASTER: Buses 1-16
- USER 1 and 2: two empty layers

Center fader section:

- DCA: DCA groups 1-16
- MAIN MATRIX: Mains 1-4 and Matrices 1-8
- AUX IN: Aux channels 1 to 8
- INPUTS: Input channels 1-40
- BUS MASTER: Buses 1-16
- USER 1 and 2: two empty layers

Right fader section:

- MAIN MATRIX: Mains 1-4 and Matrices 1-8
- DCA: DCA groups 1-16
- INPUTS: Input channels 1-40
- AUX IN: Aux channels 1-8
- BUS MASTER: Buses 1-16
- USER 1 and 2: two empty layers

WING COMPACT

WING COMPACT has a left fader section with multiple fader layers (Fig. 10) and a Main Fader Section to the right (Fig. 11).

- Left fader section:
- 1-12: Input channels 1-12
- 13-24: Input channels 13-24
- 25-36: Input channels 25-36
- 37-40 AUX IN: Input channels 37-40 and Aux channels 1-8
- BUS MASTER: Buses 1-16
- MAIN MATRIX: Mains 1-4 and Matrices 1-8
- DCA: DCA groups 1-16
- USER 1 and 2: two empty layers

The Main Fader Section consists of a 2.4" screen controlled by 4 buttons to select from Buses, DCAs, Main/Matrix and Custom Controls. 16 quick access buttons offer immediate (de)selection of any main fader and (dis)engages Sends On Faders mode where applicable. This allows extremely fast operation of all send levels.

- BUS MASTER: press any of the 16 quick access buttons to activate Sends On Faders (SOF) mode for the corresponding bus. Refer to Section 4.9 for details on SOF mode.

- DCA: press any of the 16 quick access buttons to spill the channels assigned to the corresponding DCA on the left fader section. Channels can be assigned to a DCA on the DCA Overview screen. To access it, select the DCA fader layer on the left side of the console and press the VIEW button.

  Alternatively, press the quick access button corresponding to the desired DCA, hold down the SELECT button on the Main Fader Section, navigate to the desired fader layer on the left fader section and press the SELECT button on the channels you would like to assign to the DCA. Release the SELECT button on the Main Fader Section.

  If SETUP → SURFACE → USE DCA BUTTONS 1-16 AS CC is enabled, the quick access buttons in the DCA layer can be used to assign Custom Controls.

- MAIN MATRIX: quick access buttons 1-4 activate SOF mode for the Main buses 1-4. Quick access buttons 5-8 correspond to the Custom Controls for the layer M/M. Quick access buttons 9-16 activate SOF mode for the Matrices 1-8.

- Note that the Main Sends are separate from the channel fader.

- USER: quick access buttons control the Custom Controls for the layers U1-U4.

When none of the 16 buttons are pressed, the Main Fader controls MAIN 1 by default. To change the default selection to any other MAIN, MTX or DCA, go to SETUP → SURFACE → MAIN FADER → SOURCE.

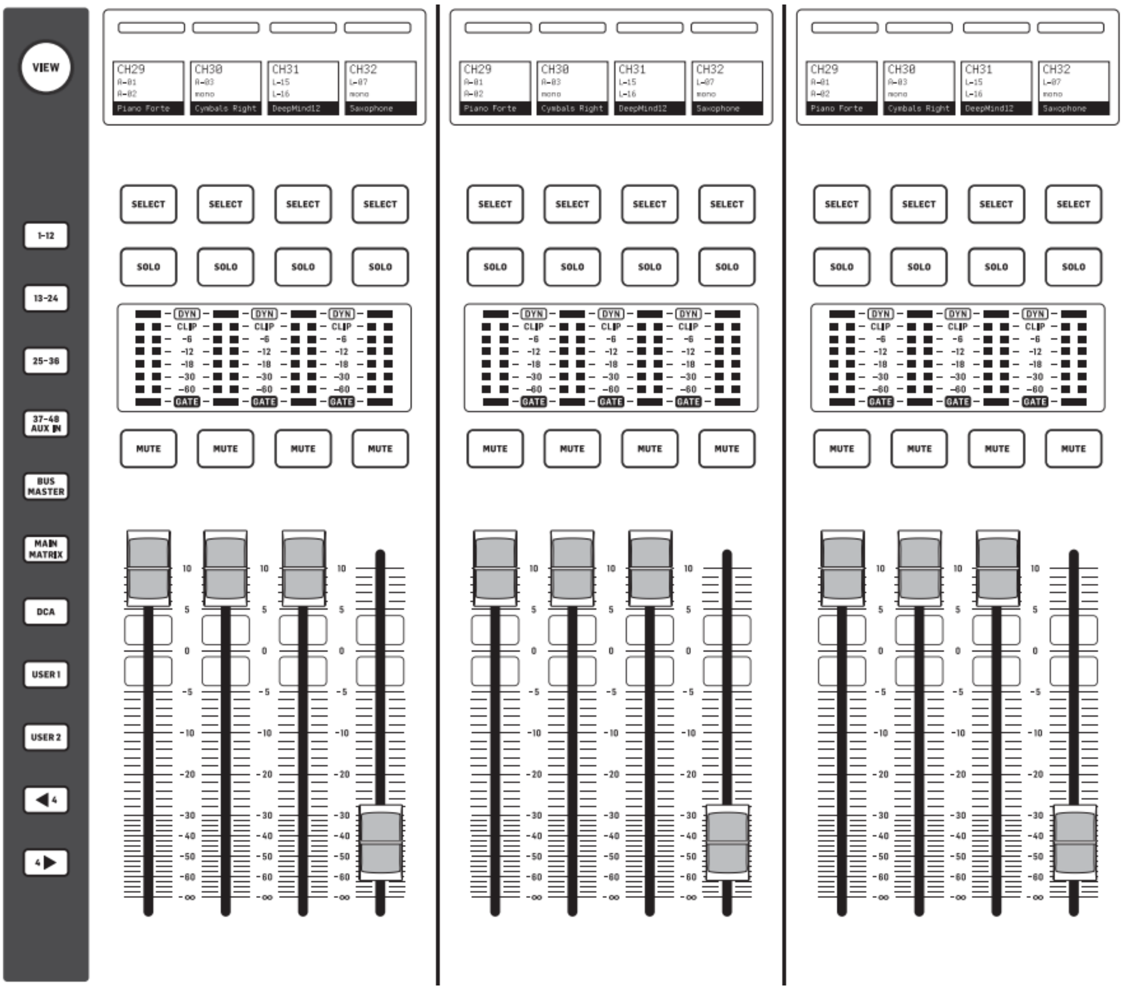

Figure 10: WING COMPACT left fader section

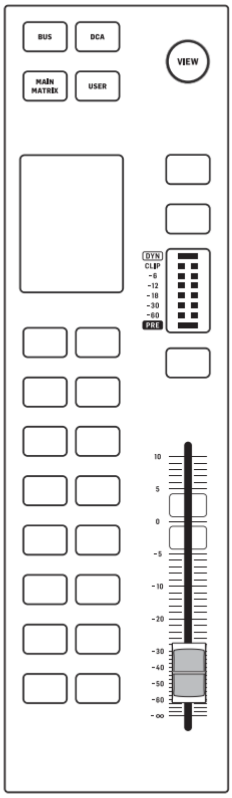

Figure 11: WING COMPACT Main Fader Section

WING RACK

WING RACK's right-side section (Fig. 12) allows access to the level, mute and solo controls for all individual channels. The section is organized vertically in pages of 4 channels and consists of a portion of the LCD (scribble strip), a rotary encoder for the channel fader and two square buttons for Mute and Solo.

Press the INPUT AUX button to toggle between the following layers:

- Input Channels 1-40
- Aux Channels 1-8

Press the BUSES MAIN button to toggle between the following layers:

- Bus 1–16
- Main 1-4 and Matrix 1-8

Press the DCA MUTE button to toggle between the following layers:

- DCA 1–16
- Mute Groups: when this layer is active, the rotary encoders are disabled, and the 8 square buttons correspond to the 8 Mute Groups.

Press the CUST TRANS button to toggle between the following layers:

- Custom Controls (see Section 4.8 for details)

- Transport: transport controls when a PLAYLIST is open or when SD cards are inserted into the WING-LIVE expansion card (see Section 8.1 for details)

Hold down the ←4 and 4→ buttons to jump to the first and last layer in each fader bank, respectively.

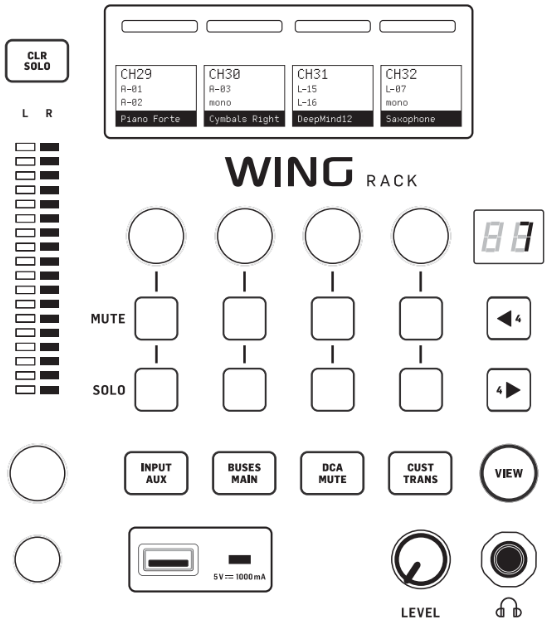

Figure 12: WING RACK 4-channel section

4.3 VIEW Buttons

Pressing one of the VIEW buttons (Fig. 13) switches the main display to a dedicated screen for the section the button was pressed on. While active, each VIEW button lights green. The tables below specify which VIEW button opens which screen.

Figure 13: VIEW button

4.4 Scribble Strips, Meters, SELECT

WING / WING COMPACT

Each fader strip features a mini display screen called a scribble strip (Fig. 14), which shows the current channel/bus number, name, and icon for quick identification. A color bar above the scribble strip provides a visual cue for related channel groups. Scribble strip details and color bar settings can be edited on the selected channel's HOME screen in the NAME/ICON/COLOR window.

Pressing SELECT a second time, while the channel is already selected, will either return the display to Channel Home, open the INS1 screen when the selected channel is a BUS, or do nothing. The behavior depends on the SEL DBL CLICK setting in the Setup → General screen.

Location	Screen opened	Screen also accessible through
USB AUDIO/DATA and MONITOR/TALKBACK	MONITORS	SETUP → MONITORS
USB AUDIO/DATA and MONITOR/TALKBACK (hold down)	USB Recorder/Player	Clicking on the USB icon on the top of every screen
Left fader section	Overview of the left fader section with the active fader layer	
Left fader section (hold down)	CONFIG	VIEW → CONFIG
Center fader section	Overview of the center fader section. The Overviews of the left and right fader sections are also accessible.	
Center fader section (hold down)	CONFIG	VIEW → CONFIG
Right fader section	Overview of the center right section. The Overviews of the left and center fader sections are also accessible.	
Right fader section (hold down)	CONFIG	
CUSTOM CONTROLS section	CUSTOM CONTROLS	SETUP → CC EDIT
Channel Strip	INPUT/TRIM and BAL/FILTER/DLY	CHANNEL HOME screen

Table 4.2: VIEW buttons on WING

Location	Screen opened	Screen also accessible through
USB AUDIO/DATA and MONITOR/TALKBACK	MONITORS	SETUP → MONITORS
USB AUDIO/DATA and MONITOR/TALKBACK (hold down)	USB Recorder/Player	Clicking on the USB icon on the top of every screen
Left fader section	Overview of the left fader section with the active fader layer	
Left fader section (hold down)	CONFIG	VIEW → CONFIG
Main fader section	FADERS	
Main fader section (hold down)	Custom Controls	SETUP → CC EDIT

Table 4.4: VIEW buttons on WING COMPACT

Location	Screen opened	Screen also accessible through
Front panel	First press: Overview screen of the fader layer that is currently active Second press: Faders screen of the fader layer that is currently active	OVERVIEW and FADERS buttons on the upper border of the main display
Front panel (hold down)	CONFIG	

Table 4.6: VIEW buttons on WING RACK

The SOLO button isolates the channel for monitoring, along with any other soloed channels or buses. The MUTE button silences the channel assigned to that strip.

Stereo level meters display input levels from -60 dB to the clip point. The Dynamics LED lights up when the dynamic processor’s threshold is exceeded, activating the compressor/expander. Similarly, the Gate LED lights when the signal drops below the noise gate’s threshold, triggering gain reduction.

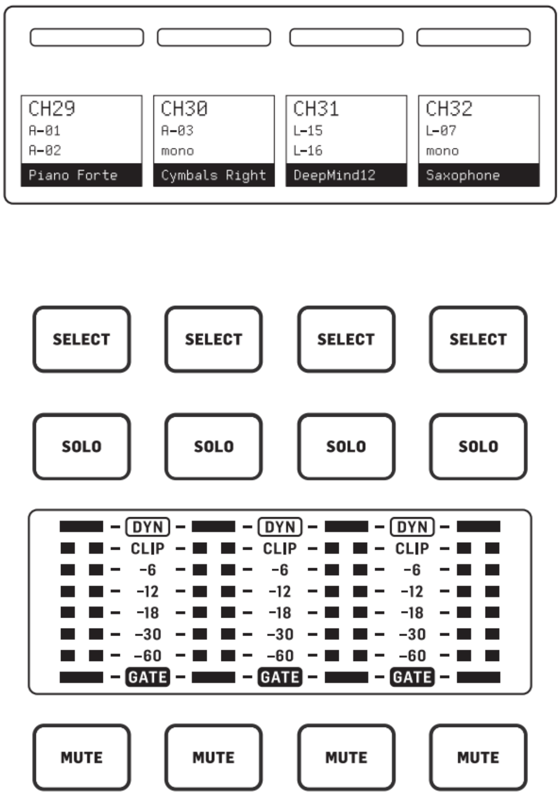

Figure 14: Scribble Strips and channel level meters

WING RACK

The LCD on the right side of WING RACK is divided into 4 sections and shows the current channel/bus number, name and icon (Fig. 12).

A color bar above the scribble strip allows quick visual identification of groups of related channels.

Scribble strip details and color bar options can be edited on the HOME screen of the selected channel, in the NAME/ICON/COLOR window.

The top button row, located below the rotary encoders, controls the mute function for the corresponding channels, while the bottom row controls the solo function. Stereo level meters in the scribble strips provide input level information, from -60 dB to the clip point.

4.5 Channel Strip Section (WING)

The channel strip (Fig. 15) provides quick access to primary parameters for the currently selected channel. The dedicated screen within this section displays details about the parameter being adjusted. LED indicators show input configuration, bus and group assignments, and level metering for convenience.

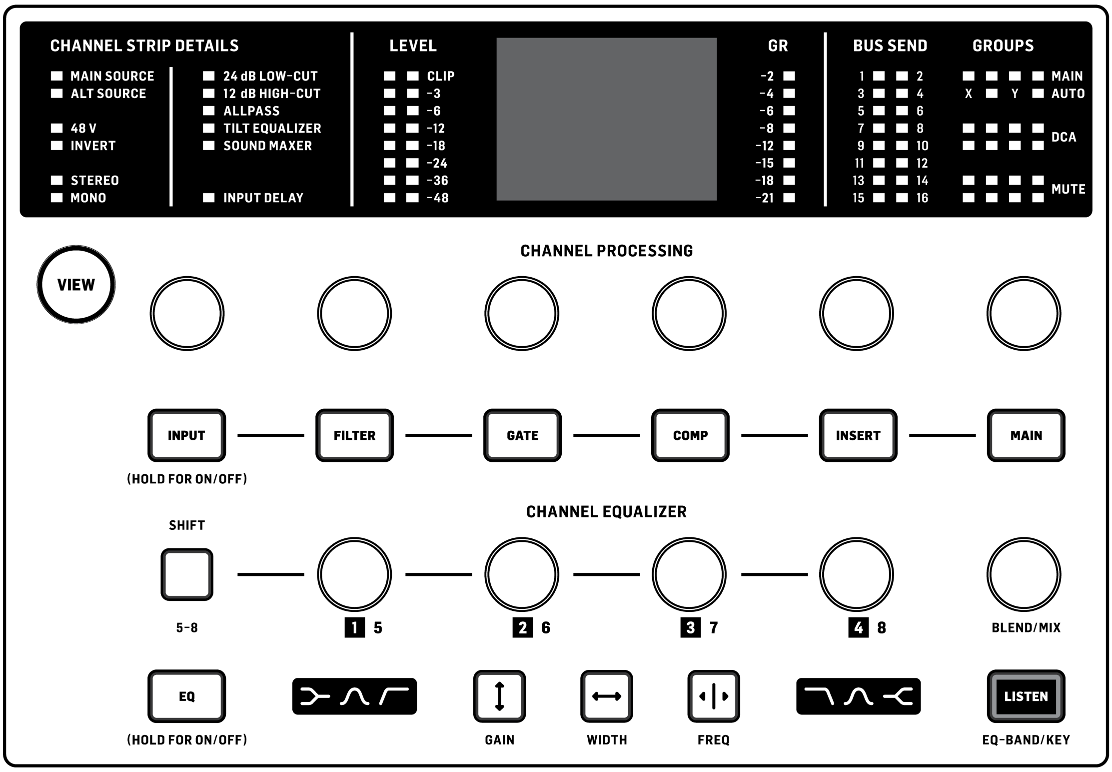

Figure 15: Channel Strip section

To display a processing slot on the channel strip screen, press the associated button or touch the capacitive encoder knob directly above. Press and hold the button corresponding to each slot to toggle the processing on or off. Once active, pressing the same button again selects another parameter. Small dots in the lower right corner of the display indicate the number of available parameters to scroll through. Adjust the selected parameter by turning the knob above each button.

The EQ section provides dedicated controls for up to six EQ bands on input channels and eight bands on buses. To activate the EQ block, press and hold the EQ button, then touch one of the four EQ encoder knobs to select a band for adjustment. Press the SHIFT button once to access low and high shelf filters or additional bands when editing a bus EQ. Use the GAIN, WIDTH, and FREQ buttons to select which element of the current band will be adjusted with the knob.

Press the LISTEN button in the lower right corner to monitor the selected EQ band in isolation.

The BLEND/MIX knob serves as a wet/dry adjustment for the EQ block. When set above 100%, it intensifies the current EQ setting by modifying each band's gain according to the selected percentage.

4.6 4-Channel Section (WING)

The 4-Channel Section (Fig. 16), located at the top-right corner of the console, provides dedicated controls for a selected bank of four channels. Pressing one of the 8 buttons at the top of the section activates the 4 knobs and 4 buttons below the scribble strips, allowing channel adjustments without selecting the channel for editing.

This section operates independently from the main control surface, enabling a second user to work on the console simultaneously. Additionally, the F1-F3 controls can be assigned to any Custom Control function via SETUP → SURFACE.

Layer	Knob	Button
GAIN / 48V	Preamp gain	Phantom power on/off
GATE	Gate threshold	Gain on/off
COMP	Compressor threshold	Compressor on/off
FILTER	Low-cut filter cutoff frequency (press again for High Cut)	Low-cut filter on/off
PAN / MAIN	Panning (press again for WIDTH)	MAIN1 send on/off
F1-F3	Send level to the selected BUS (hold down F1-F3 knob to change BUS)	Send on/off

Table 4.8: Controls in the 4-Channel Section

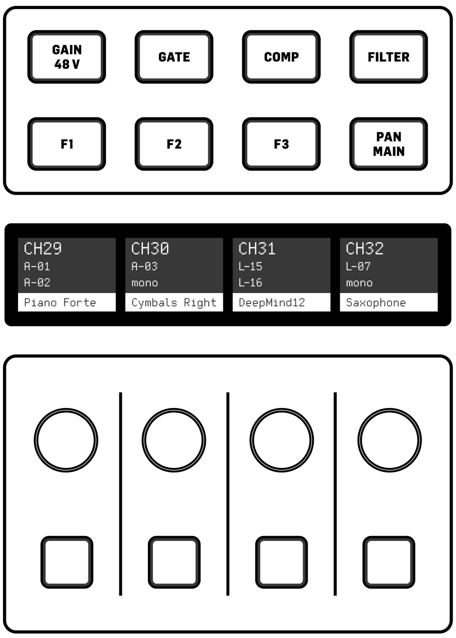

Figure 16: 4-Channel Section

4.7 USB Audio/Data and Monitor/Talkback (WING / WING COMPACT)

The USB type-A connector allows using a USB flash drive to save and load data such as snapshots, snippets, and channel presets. It also supports recording and playback of 2 or 4-track WAV audio files and can charge portable devices like phones or tablets. Flash drives can be safely disconnected when the amber LED is off.

Dedicated knobs control the PHONES (headphones) and MONITOR (speakers) output levels on WING and WING COMPACT. Both signals can be routed to any physical output. On WING RACK, only a dedicated PHONES LEVEL knob is available (Fig. 12). The headphone jacks on WING, WING COMPACT, and the front of WING RACK correspond to the signals sent to AUX OUT 7 and 8.

The DIM button temporarily attenuates monitoring volume and holding it down mutes the monitor. The MONO button sums the monitor signal to mono, while holding it down swaps the left and right signals. Blinking buttons indicate active mute or swap functions—press again to disengage.

The TALK LEVEL knob adjusts the talkback microphone volume, and the TALK A and B buttons send the talkback signal to designated destinations, as configured in the MONITORS section.

Press the VIEW button to access monitoring settings, including DIM attenuation, talkback mic routing, and other parameters. For more details, see Chapter 7 Monitors.

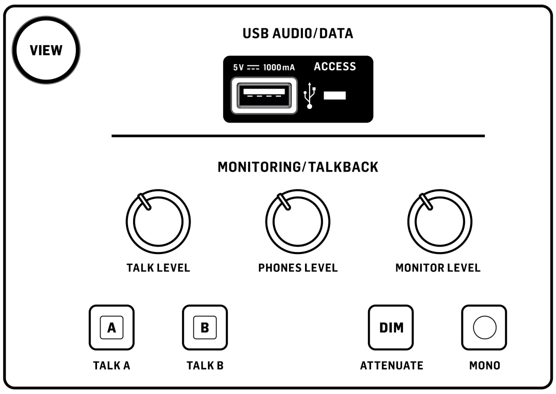

Figure 17: USB AUDIO/DATA and MONITOR/TALKBACK section (WING)

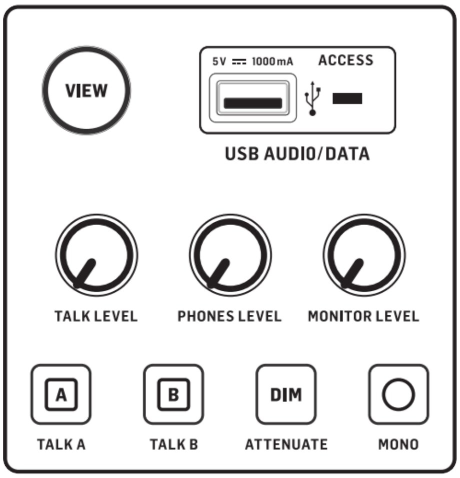

Figure 18: USB AUDIO/DATA and MONITOR/TALKBACK section (WING COMPACT)

4.8 CUSTOM CONTROLS

The CUSTOM CONTROLS (CC) section (Fig. 19) allows you to assign various parameters to the rotary encoders and buttons for quick access.

The number of available hardware CC buttons and rotary encoders varies by WING model. However, all layers are configurable on any model, allowing you to prepare your CC configuration on a different WING model if necessary.

Figure 19: Custom Controls screen

To assign a parameter to a CUSTOM CONTROL knob or button, go to SETUP → CC EDIT or press the SETUP button twice.

Alternatively, press the VIEW button on WING's CC section, CUST TRANS layer on WING RACK, or hold down the VIEW button on WING COMPACT's Main Fader Section.

The following layers are available for editing on all consoles:

- 1-16: layers 1-16 with 4 rotary encoders and 8 buttons each (WING and WING RACK)
- U1-U4: User layer buttons 1-16 (WING COMPACT)
- M/M: Main Matrix layer buttons 5-8 (WING COMPACT)
- D1-D4: DCA layer buttons 1-16 (WING COMPACT)

Different parameters can be assigned to rotary encoders and buttons. It is also possible to assign parameters with more than two states (on/off) or continuously adjustable parameters to a button (e.g., a reverb's pre-delay time).

To adjust a parameter with more than two states using a button, hold down the button and turn the knob directly above it (WING and WING RACK). On WING COMPACT, hold down the custom control and use the knob to the right of the main display.

Custom Naming and Coloring

Each CC button can have a custom name and color (ensure COLOR → LIGHT is enabled).

On WING and WING RACK, the color is assigned to the scribble strip above a rotary encoder and two buttons.

On WING COMPACT, each CC has an individual color shown on the Main Fader Section display.

Special Name Functions

Insert an asterisk (*) directly before a CC's name (without a space) to display its exact value on the screen alongside the CC's name.

Insert a vertical bar (|) directly before a CC's name (without a space) to invert the background illumination behind the CC's name.

WING

The CC section is organized into two parts (Fig. 20): the top part is empty by default and the lower part contains a predetermined set of controls, as well as an additional User layer.

The top section consists of four rotary encoders and eight buttons arranged across 16 layers that can be navigated with the up and down buttons. This configuration provides a total of 64 assignable rotary encoders and 128 assignable buttons.

The bottom part contains the following default layers:

- 2-TRACK TRANSP: controls the 2 or 4-track USB recorder transport
- MULTI TRANSP: controls the WING-LIVE multitrack recorder transport
- MUTE GROUPS: switches the 8 Mute Groups on and off
- SHOW CONTROL: controls Scene recall and navigation when a Show file is active
- USER: empty layer for additional user-assignable parameters

The navigation arrows at the bottom of the CC section can be used to select parameters on screen, and the scroll wheel can be used to adjust the selected parameter.

WING COMPACT

The custom controls are organized into three layers (Fig. 11):

- USER layer consisting of 16 user-assignable buttons
- MAIN MTX layer buttons 5-8 (unused by default and available as CCs)
- DCA layer buttons 1-16 (when SETUP → SURFACE → USE DCA BUTTONS 1-16 AS CC is enabled)

WING RACK

The CUST TRANS layer offers 16 layers with 4 rotary encoders and 8 buttons each, following the same arrangement as the top half of the CC section on WING (Fig. 12).

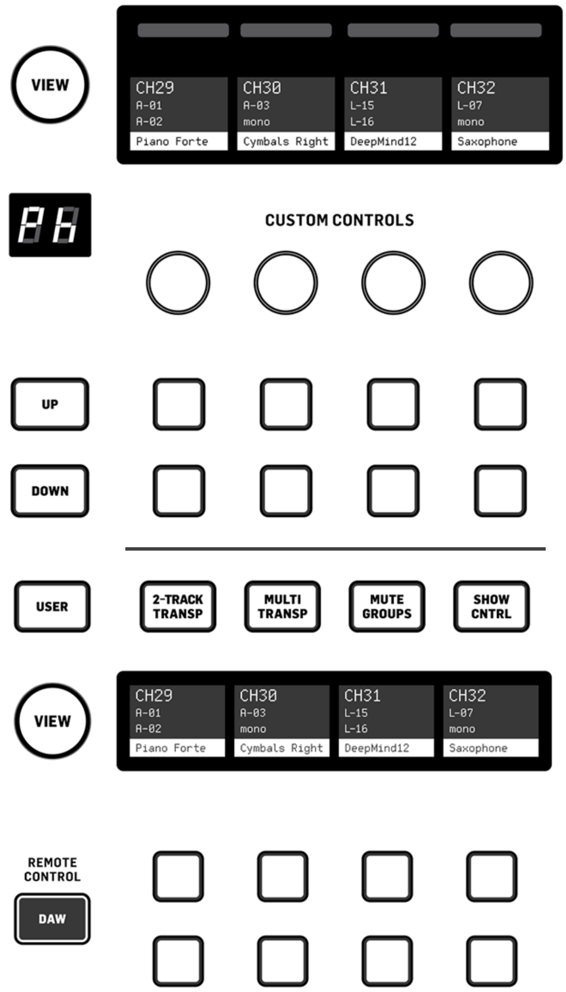

Figure 20: Custom Control Section (WING)

DAW Control

When the DAW button is pressed on WING to switch to DAW Remote Control, the console's surface takes control of the DAW (if SETUP → DAW ENABLE is set to on). In this mode, the wheel adjusts the cursor position. The lower part of the CC section consistently controls multiple DAW parameters. The upper part of the CC section can either display the assigned custom controls or control DAW parameters, depending on the SETUP → DAW → USE UPPER CC FOR DAW setting."

On WING COMPACT and RACK, the DAW LAYER button must be assigned to a CUSTOM CONTROL: CC EDIT → FUNCTION → OTHER → DAW LAYER.

Please refer to the DAW Control Guide on behringer.com for more information on how to set up remote control with different DAWs.

GPIO

Click on the GPIO button at the top of the CC EDIT screen to assign General-Purpose Inputs/Outputs (GPIOs). GPIOs can be used to send or receive control signals to hardware, such as footswitches. GPIOs can be set up in three modes: Toggle (TGL), Input (IN), and Output (OUT). Each mode has a Normally Open (N.O.) and a Normally Closed (N.C.) configuration, which define the default state of the circuit when no external action is taken.

Toggle Mode (TGL)

Toggle mode changes the GPIO state from HIGH (1) to LOW (0) or vice versa. It can be used to control external devices from the console.

- TGL N.O.: The circuit is open by default (no current flows). The first toggle closes the circuit, and the next toggle opens it again.
- TGL N.C.: The circuit is closed by default (current flows). The first toggle opens the circuit, and the next toggle closes it again.

Input Mode (IN)

Input mode is used to control console functions with external devices such as foot switches.

- IN N.O.: The circuit is open by default (no current flows), and the GPIO reads LOW (or 0). When the external switch is pressed, the circuit closes, and the GPIO reads HIGH (or 1).
- IN N.C.: The circuit is closed by default (current flows), and the GPIO reads HIGH (or 1). When the external switch is pressed, the circuit opens, and the GPIO reads LOW (or 0).

Output Mode (OUT)

Output mode is used to control external devices from the console.

- OUT N.O.: The circuit is open by default (no current flows), and the external device is off. When the assigned action is triggered in the console, the GPIO pin outputs HIGH, the circuit closes, and the external device turns on.
- OUT N.C.: The circuit is closed by default (current flows), and the external device is on. When the assigned action is triggered in the console, the GPIO pin outputs LOW, the circuit opens, and the external device turns off.

4.9 Sends On Faders Mode

WING

Sends On Faders (SOF) mode is available to quickly adjust the channels' send levels to a particular bus. Follow these steps to adjust the send levels from multiple channels to a Bus using SOF mode:

1. Select the desired Bus, Main or Matrix in the center or right-fader section.
2. Press the SOF FLIP button.
3. The Send levels from the individual channels can be controlled with the corresponding channel faders. Unmuting the channels activates each send to the selected Bus, Main or Matrix.

The sends of a single channel to multiple Buses, Mains or Matrices can also be adjusted with SOF mode. Repeat the steps above and select the desired Input or Aux channel. The sends from this channel to the Buses, Mains and Matrices are shown on the corresponding Bus/Main/MTX channel on the other fader sections.

When Alternative SOF Mode is enabled in the SETUP → SURFACE screen, two different SOF modes can be used: the standard SOF mode whose destination changes according to the channel that is selected and the Alternative SOF Mode in which the destination remains fixed even if another channel is selected.

Activate the standard SOF mode by selecting the desired channel and holding the SOF FLIP button.

Activate the Alternative SOF by holding the SELECT button on the desired channel, Bus, Main or Matrix. Exit both SOF modes by pressing the SOF FLIP button.

WING COMPACT

WING COMPACT's Main Fader Section is optimized for SOF mode. When pressing one of the 16 buttons in this section, SOF mode is activated with the selected bus as destination. The left fader section then controls the sends from its corresponding channel to the selected bus.

WING RACK

1. SOF mode can be activated on WING RACK as follows:
2. Select a destination bus on CUSTOM CONTROLS → FUNCTION: SENDS ON FDR → CHANNEL.
3. Press the Custom Control.
4. Navigate to the desired fader bank.

The send levels from the channels shown in the scribble strips to the assigned bus can now be controlled with the corresponding rotary encoders. Unmuting the corresponding channels activates each send to the selected BUS.

The SENDS screen also serves a similar purpose. See Section 5.9 for details.

The sends of a single channel to multiple BUSES can also be adjusted with SOF mode. To do so:

1. Select an Input or Aux channel on Custom Controls → FUNCTION: SENDS ON FDR → CHANNEL.
2. Press the Custom Control.
3. Navigate to the BUSES MAIN fader bank.

Section 5

Main Screens

5.1 HOME

Channel Home

Pressing the HOME button opens the Channel Home screen (Fig. 21) where you have access to all the processing slots, customization options and settings for the selected channel. Note that the processing slots available change, depending on whether an input channel, Aux, Bus, Matrix or Main is selected.

Figure 21: Channel Home screen

On WING RACK, pressing the HOME button opens the FADERS screen (Fig. 56) and includes the left toolbar also found in the MONITOR screen (Fig. 72). Press the HOME button again to toggle to the OVERVIEW screen (Fig. 53). Click on the top section of a fader to access its Channel Home screen.

Clicking on the channel number on the top left corner opens the Channel Select screen which shows all 40 input channels, 8 Aux channels, 16 Buses, 4 Mains and 8 Matrices. Clicking on any of them opens the corresponding Channel Home screen. The information shown on the Channel Home screen also depends on the channel type. If an input or Aux channel is selected, the Source, Trim and small graphic representations of the active signal processors are shown in the center of the screen. You can click on them to go directly to each slot's dedicated screen.

If a Bus, Matrix or Main is selected, a grid with the channels, Aux channels, Buses or Main currently assigned to the selected Bus, Matrix or Main is shown. Clicking on this grid opens the Bus Feed Configuration screen (explained below in this section).

Icon / Color

This tab allows you to select the icon and color to be displayed on the channel's scribble strip. The light bar above the scribble strip can be turned on or off.

LINK CUSTOMIZATION TO SOURCE syncs the name, icon and color of the Source and the channel it is assigned to. When this option is disabled, it is possible to transfer the customization options from the Source to the channel and vice versa by dragging the Channel or Source box in the desired direction in the Copy Customization section. This section is available when assigning a Source in the Channel Home (Fig. 21), in the the Input slot (Fig. 62) and in the Routing → Channels screen (Fig. 30). The customization link can be activated or deactivated for all input and Aux channels by clicking on NONE or ALL.

Name

Input the desired name for the channel. After an icon has been selected, multiple name suggestions are displayed here.

Tags

Use the TAGS tab to assign a channel, Aux channel, Bus, Matrix or Main to a DCA or Mute Group. There are 16 available DCA groups, labeled D1 to D16, and 8 Mute Groups.

The "TALK ON" tag unmutes the channels with this tag when talkback is activated. The "TALK SOLO" tag activates Solo on the channels with this tag when talkback is activated. The tags that indicate "TALK A" or "TALK B" react only when the corresponding TALK A or TALK B button is pressed. The tags "TALK" that do not specify "A" or "B" react to both talkback buttons.

Initalize / Copy

The INIT and COPY buttons are shown in all the channel's processing slots. The parameter SCOPE that will be initialized or copied is preselected to match the processing slot that was open when clicking on INIT/COPY. If the Channel Home screen is open, all the channel's parameters, except its SOURCE assignment (CONN) are preselected.

Figure 22: INIT screen

In the COPY screen (Fig. 23), the channel to be copied is shown in the top left corner. Click on it to select another channel. The destination channel or channels can be selected in the grid labeled Destination. Parameters can be copied between different channel types (input CHAN, AUX, BUS, MAIN or MATRIX). Only the common parameters between the original channel and the destination are copied.

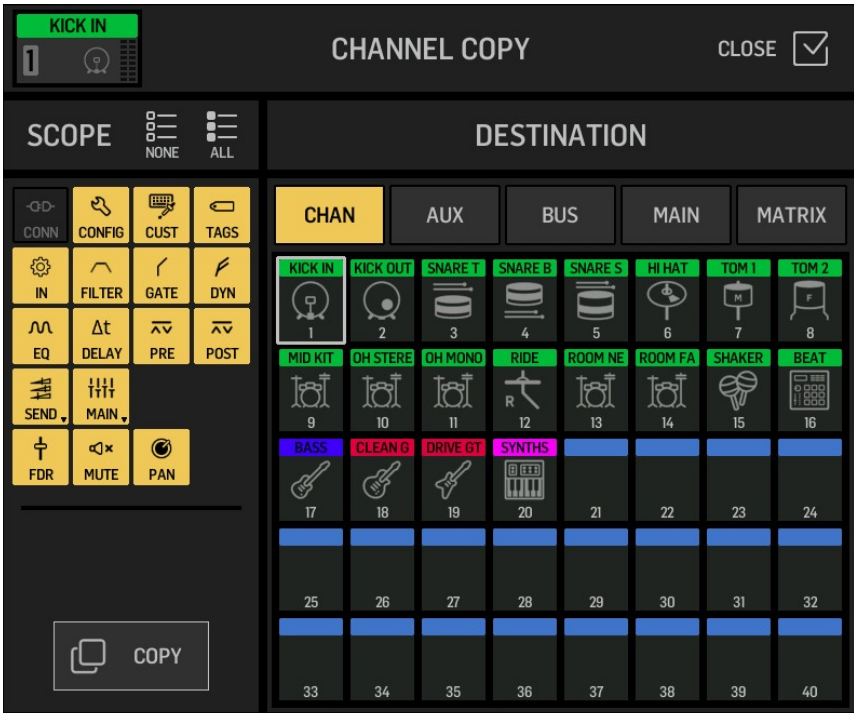

Figure 23: COPY screen

RTA and EQ/RTA Settings

When on the Channel Home screen, the Settings button on the top right corner opens the RTA Settings (Fig. 24) for the spectrum analyzer shown in the Channel Home tab.

- RTA Source
  - IN: raw Source assigned to the channel
  - EQ: post EQ signal
  - POST: post fader signal
- RTA Range: the range of the vertical axis shown on the RTA
- RTA Decay: the rate at which each bar in the RTA drops back to the minimum value
- RTA Detector: use Peak for immediate measurements, and RMS or AVG (average) for a slower measurement
- RTA Autogain adjusts the vertical scale of the RTA graph so that the signals are centered vertically in the graph.
- Fixed Gain shifts the vertical scale of the RTA graph by a specific value for manual adjustments.
- Size/Mode determines how much space the RTA covers in the Overview screen EQ graph or turns it off altogether.
- Color: three color options (red, amber, blue) and multiple opacity settings (25%, 50%, 75%) for the RTA

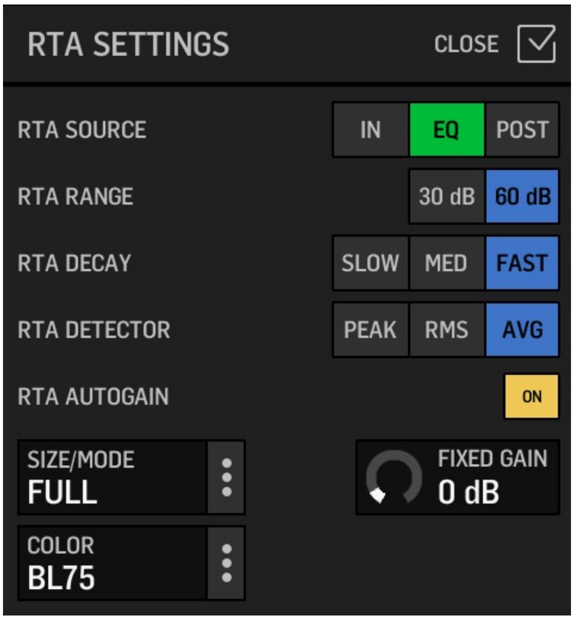

Figure 24: RTA Settings screen

When the EQ processing slot is selected, the SETTINGS button on the top right corner opens the EQ/RTA Settings (Fig. 25) which adds the option to change the low shelving and high shelving filters to parametric filters.

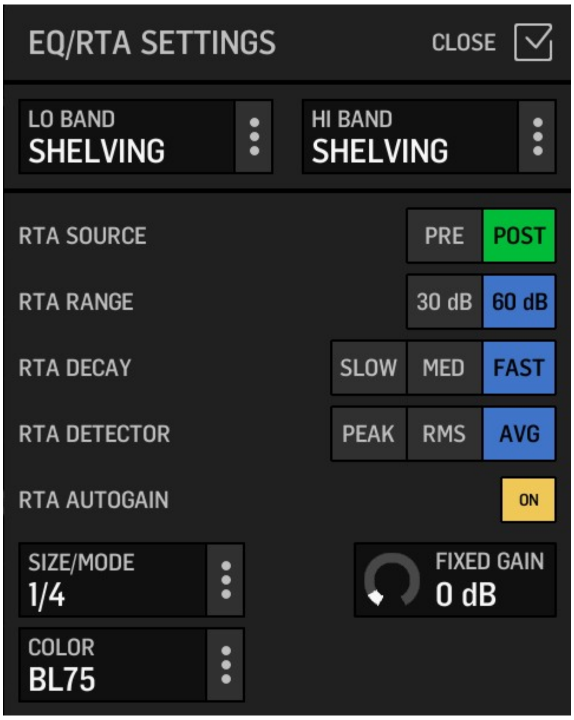

Figure 25: EQ/RTA SETTINGS screen

Bus Feed Configuration

Go to the HOME screen of the desired Bus, Matrix or Main and click on the Channel/Bus/Aux grid in the center of the screen (Fig. 26) to open the Bus/Matrix/Main Feed Configuration screen (Fig. 27). This grid shows the Input channels, Aux channels and Buses currently being sent to the selected bus.

Alternatively, go to the second processing block in a Bus and click on the Bus Feed section.

When a Send is disabled, the corresponding cell in the grid is greyed out.

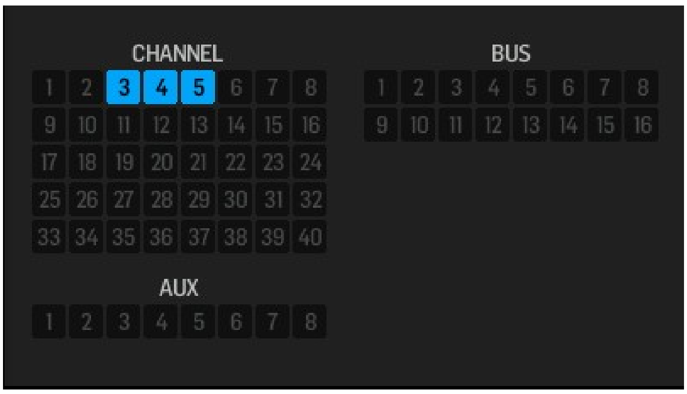

Figure 26: Bus Feed overview grid

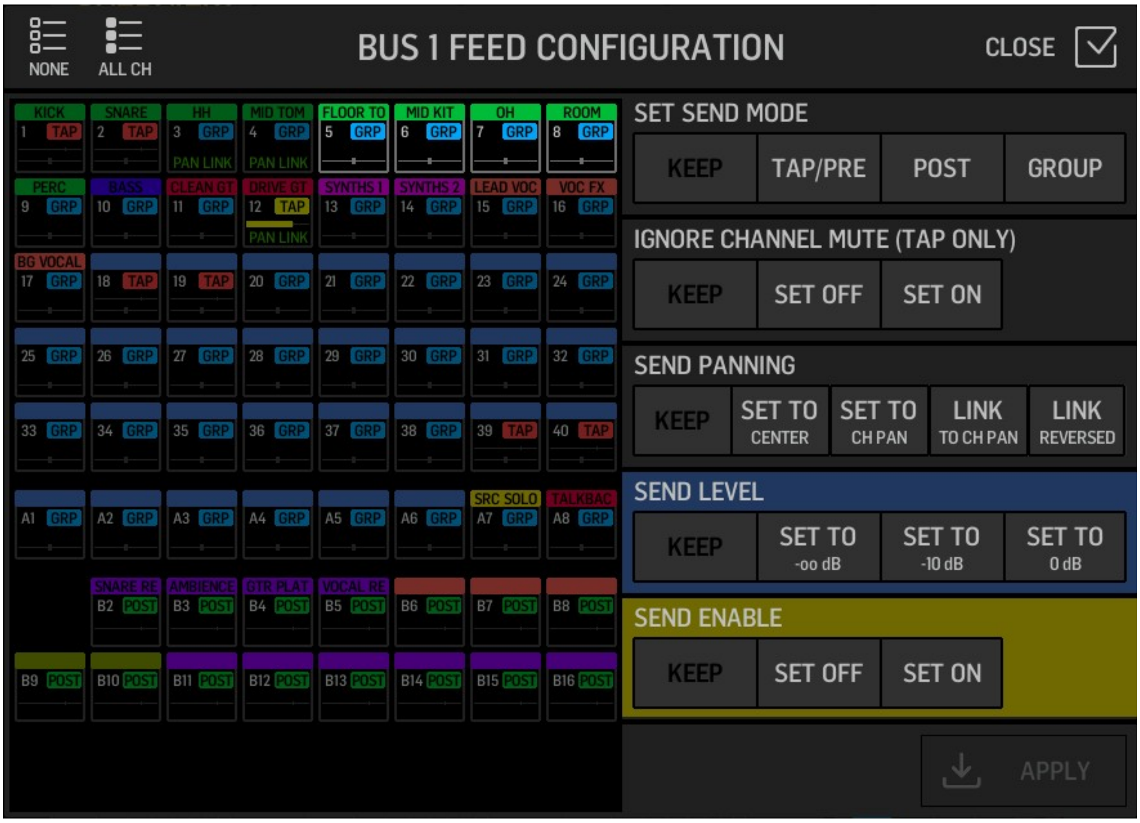

Figure 27: Bus Feed Configuration

The Bus Feed Configuration screen allows you to quickly send Input and Aux channels, Buses, Mains or Matrices to the selected Bus, Matrix or Main. The following settings can be applied to the selected channels:

- Send Mode:
  - KEEP: keep the existing setting
  - TAP/PRE: send the send mode to TAP point (for Input and Aux channels) or pre-fader (for Bus, Main or Matrix)
  - Post: set the send mode to post-fader
  - Group: set the send mode to Group mode, disabling the send level.
- Ignore Channel Mute (TAP only): this setting applies only to sends in Input and Aux channels in TAP mode
  - Keep: keep the existing setting
  - Set Off: disable the send when the source channel is muted
  - Set On: keep the send active even when the source channel is muted
- Send Panning:
  - Keep: keep the existing setting
  - Center: set the send panning to the center position

  - CH Pan: set the send panning to the same position as the channel panning. The send panning can subsequently be adjusted independently.
  - Pan Link: link the send and channel panning
- Send Enable:
  - Keep: keep the existing setting
  - Set Off: disable the send
  - Set On: enable the send
- Send Level:
  - Keep: keep the existing setting
  - $-\infty$ dB: set the SEND LEVEL to minimum
  - -10 dB: set the SEND LEVEL to -10 dB
  - 0dB: set the SEND LEVEL to 0 dB

Click on APPLY after doing the desired changes.

5.2 EFFECTS

The EFFECTS screen (Fig. 28) provides an overview of the 16 effect units that can be loaded into the FX Rack and used as insert points. Users can choose from a wide selection of analog emulations, adjust parameters, and monitor the output level of each effect.

Please note that the analog emulations and digital effects available in the GATE, EQ and COMP slots of each channel strip are loaded directly onto each channel, rather than occupying the 16 slots on the EFFECTS screen.

Effects loaded in the GATE, EQ and COMP on each channel do not introduce additional latency. In contrast, effects loaded through inserts points do add latency. For more information, please refer to section 6.6.

The eight **PREMIUM FX** slots can be used to load effects from any of the three categories:

- FX1-8 (time-based effects)
- STD (equalizers, amp simulations, enhancers, etc.)
- CH (channel strips)

FX slots 1-8 provide access to effects that require external memory—such as those with delay lines—not available in slots 9-16. These slots are ideal for high-quality reverbs and creative delay effects.

Channel strips combine three effect devices into a single slot (four in the case of MASTERING).

The eight **Standard FX** slots can accommodate only the effects from the STD and CH categories.

Effects are typically applied to channels in one of two ways: via Bus Sends or insert points. For example, reverbs are usually configured through Bus Sends, while an amp simulation is typically

Figure 28: EFFECTS screen

used as an insert point.

To set up an effect through a **Bus Send**:

1. Load the desired effect into the insert point of a Bus (see Section 6.3).
2. Ensure that this channel is routed to the Main bus in the Main Sends section (see Fig. 1).
3. Send the desired channels to the Bus where the effect was loaded with one of these methods:

- Using SOF mode (see Section 4.3)
- In the Bus Feed Configuration (as discussed earlier in this section)
- On the channel's Sends page (covered later in this section)

To set up an effect as an **Insert Point**:

1. Load the desired effect into a slot in the FX Rack.
2. Open the insert point of the target channel and select the desired FX Rack slot.

5.3 METERS

The METERS screen (Fig. 29) displays pre-fader level meters and mute status for all signal paths on the console. The level meters are organized into the following groups:

- 40 Input channels
- 8 Aux channels
- 16 Buses

- 16 DCAs
- 4 Mains
- 8 Matrices

Touching any group opens the FADERS screen for the corresponding section.

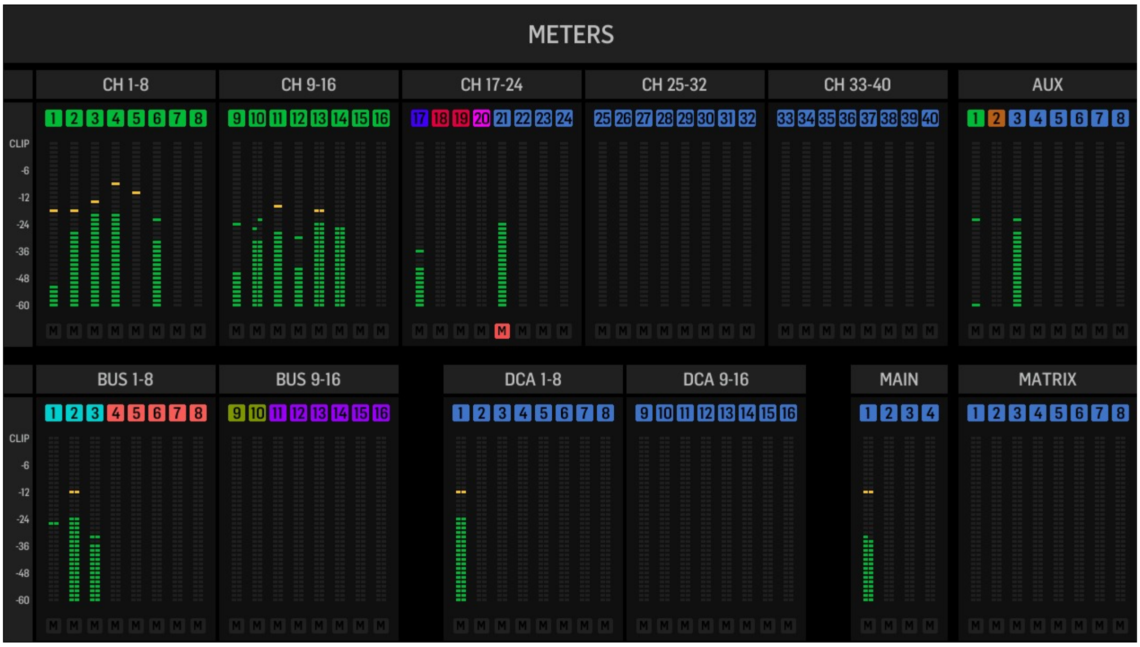

Figure 29: METERS screen

5.4 ROUTING

The patching of Sources and outputs is configured on the three Routing screens: CHANNELS, SOURCES and OUTPUTS. Three buttons on the upper border allow users to toggle between the three screens. Alternatively, the same adjustments can also be made on each channel's Input section (see Section 6.1).

CHANNELS

The CHANNELS screen (Fig. 30) is ideal for quickly patching multiple Sources to the desired channels. The left panel displays the 40 Input channels and 8 Aux channels, while the right panel shows the available Sources that can be assigned to the 48 channels.

The Channel Input buttons control whether the channel's MAIN or ALT input is configured. This control is also available on each individual Channel Input section (see Fig. 62). This setup follows the concept of an inline console, allowing a single channel to have two inputs permanently patched and switched according to the user's needs.

The Copy Customization buttons work as explained in section 5.1, Icon / Color, when "Link Customization to Source" is disabled.

Figure 30: CHANNELS screen

To assign a Source to a channel:

1. Click on the desired channel.
2. Click on the Source you want to assign to that channel.
3. Ensure that the lock icon is green before editing. If the lock icon is red, editing is not possible.

The +1 AUTO button automatically selects the next channel after a Source is assigned, making it quicker to patch multiple channels.

Clicking on the SOURCE GROUP dropdown menu displays the available Sources that can be assigned to the 40 input channels and 8 Aux channels:

- Local In: on-board microphone preamps (XLR F)
- Aux In: on-board line-level inputs (TRS)
- AES/EBU In: stereo AES/EBU input
- AES50 A/B/C: 3 AES50 ports (48 inputs each)
- USB Audio: 48 channel USB 2.0 audio interface
- Exp Card: up to 64 inputs through an expansion card (WING-LIVE, WING-DANTE, WING-MADI)
- Module: up to 64 inputs through the internal module (Waves SoundGrid or Dante)
- Bus: 16 stereo Buses
- Main: 4 stereo Main buses

- Matrix: 8 stereo Matrix buses
- Oscillator: two independent oscillators (sine wave, pink noise, white noise)
- ST Connect: up to 32 inputs through the StageConnect port
- USB Player: up to 4 channels from the USB player (Type A port) on the front panel. Files must be uncompressed .wav audio.
- User Signal: up to 48 signals that can be derived from different points in the signal flow of Input channels, Aux channels, Buses, Mains or Matrices; as well as repatched external Sources if using User Patches. See more details below.

SOURCES

The SOURCES screen (Fig. 31) allows you to determine the Sources' properties such as mono, stereo or mid/side configuration, polarity, gain, phantom power, preamp remote control, customization, etc. The left side of the screen shows the Source Groups. After selecting a Source, the right side of the screen shows its properties.

Touch and drag to select multiple Sources. Parameter changes are applied to all selected Sources.

Figure 31: SOURCES screen

To assign a stereo or mid/side Source to a single channel, two input channels must be grouped as a single stereo or mid/side Source in the right panel.

Each Source can be renamed by clicking on the RENAME icon. The Settings, Icon/Color and Tags provide further settings. Sources can also be assigned to Mute Groups in the Tags tab.

AES50 and Local In Sources have additional options for customization and remote control. REMOTE enables the selected Source's preamp settings to be controlled by the console connected to the selected AES50 port.

To remote-control a preamp, the Source must be routed from the remote-controlled device to the remote-controlling device via AES50 and ENABLE HA REMOTE must be enabled on the remote-controlled device's AES50 port through which the connection has been established (see Section 5.5).

RCV CUST (receive customization) allows the console to receive the customization of the Source defined in the transmitting console, even when remote control is not enabled.

Hold down the desired REMOTE or RCV CUST button to activate or deactivate remote control or customization transfer for all the Sources in the current AES50 port.

OUTPUTS

The OUTPUTS screen (Fig. 32) allows you to route any Source to any digital or analog output. The left side of the screen shows the available outputs. All Output Groups are accessible on the dropdown menu at the top of the screen.

The signal fed into the selected output is chosen on the right side of the screen. All the Source Groups are accessible on the dropdown menu at the top of the screen.

The Source Group "MONITOR" has a PHONES and SPEAKERS output that can be routed to any available output. The specific signal sent to both Monitor outputs is chosen in the MONITORS screen (see Chapter 7).

Figure 32: OUTPUTS screen

5.5 SETUP

All the console's settings are organized in 5 tabs in the Setup screen: General, Audio, Surface, Remote and DAW. Note that some parameters vary, depending on the specific WING model.

On the top right corner, the **INITIALIZE CONSOLE** and **SHUTDOWN** buttons are always visible. After clicking on INITIALIZE CONSOLE, it's possible to select the specific parameters that are going to be initialized (Scope). Note that parameters selected in Global Safe are be overwritten by the Initialize Console function.

After clicking on SHUTDOWN, a message will appear: "IT IS SAFE TO SWITCH DOWN THE CONSOLE NOW". After this message appears, it's possible to restart the console by holding down the EFFECTS button and then pressing the HOME button.

The **CONSOLE OPTIONS** screen is only available if the UTILITY button is held down while turning on the console. The CONSOLE OPTIONS dialog accessible through this button is used for registering the internal Audio-over-IP (AoIP) module. Detailed instructions are provided in the respective module's Quick Start Guide.

GENERAL

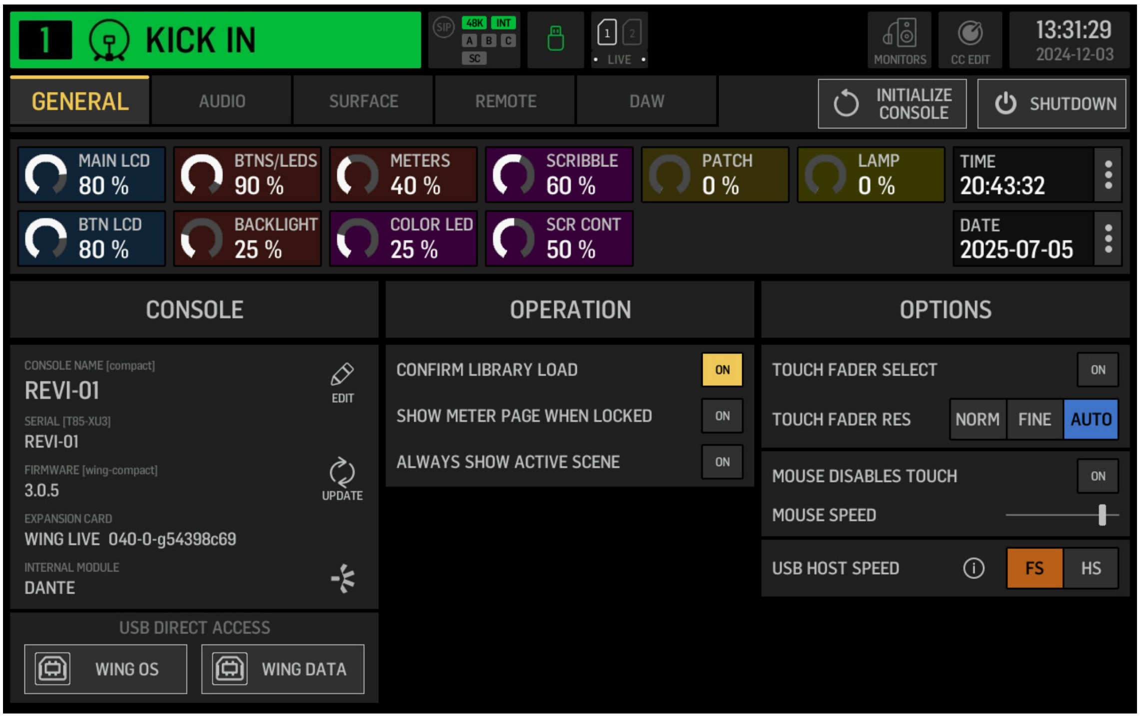

Figure 33: SETUP → GENERAL

The top part of the screen controls multiple lighting options:

- Brightness
  - MAIN LCD: main display
  - CH STRIP: channel strip section display
  - BTNS/LEDS: orange-lit buttons
  - BACKLIGHT: white-lit buttons
  - METERS: LED meters
  - COLOR LED: scribble strips' color LEDs
  - SCRIBBLE: scribble strips

- Contrast
  - Scribble strips' contrast
- Lamps
  - PATCH: rear panel lighting
  - GLOW: bottom inner panel lighting
  - LAMP: external lamp intensity

The TIME and DATE can be set on the right side of the screen. It is possible to choose between 12 hours and 24 hours format, as well as between YYYY-MM-DD and DD-MM-YYYY formats.

The bottom part of the screen is organized into three sections:

- CONSOLE
  - EDIT: enter the desired console name. This name is shown on the remote-control apps.
  - UPDATE: select a firmware file stored in the USB stick inserted on the front panel.
  - WING OS: changes the USB 2.0 connection to USB Disk mode for firmware updates.
  - WING DATA: changes the USB 2.0 connection to USB Disk mode for preset transfer (WAV files, snippets, snapshots, FX presets, etc.)

    Note: when WING OS or WING DATA mode is active, eject the disk from the computer before disconnecting.
- OPERATION
  - CONFIRM LIBRARY LOAD: activates a confirmation dialog when loading preset files in the LIBRARY screen.
  - SHOW METER PAGE WHEN LOCKED: displays the METERS screen when the main display is locked (by holding down HOME or HOME + any button next to the main display)
  - ALWAYS SHOW ACTIVE SCENE: shows the currently active SCENE on the top left corner (snapshot, snippet, FX preset, etc.)
- OPTIONS
  - USE CRSR/WHEEL: enables using the large wheel below the Custom Controls section to adjust the parameter selected on the main display. The parameter can also be selected with the 4 arrow buttons.
  - TOUCH FADER SELECT: allows you to select a fader on the screen and use the encoder to the right of the main display to adjust it.
  - TOUCH FADER RES: adjusts the channel fader resolution on the touch screen. In AUTO mode, dragging the fader normally activates NORM mode and dragging outside of the touch area activates FINE mode.
  - MOUSE DISABLES TOUCH: disables the touch screen when a mouse is connected to the front USB port.

- USB HOST SPEED
  * FS: USB recording on the front port is restricted to 2 channels and 16 bits (should only be used if the write speed on the USB device is too slow for 4-channel / 24-bit recording).
  * HS: enables 4-track and 24-bit recording on the front port.
- NETWORK CONFIG (only available when the AoIP-WSG module is installed)
  * Switched: both Ethernet ports transmit audio (to/from the internal AoIP module) and WING remote control data
  * Separated: the left Ethernet port transmits WING remote control data, and the right Ethernet port transmits audio (to/from the internal AoIP module)

AUDIO

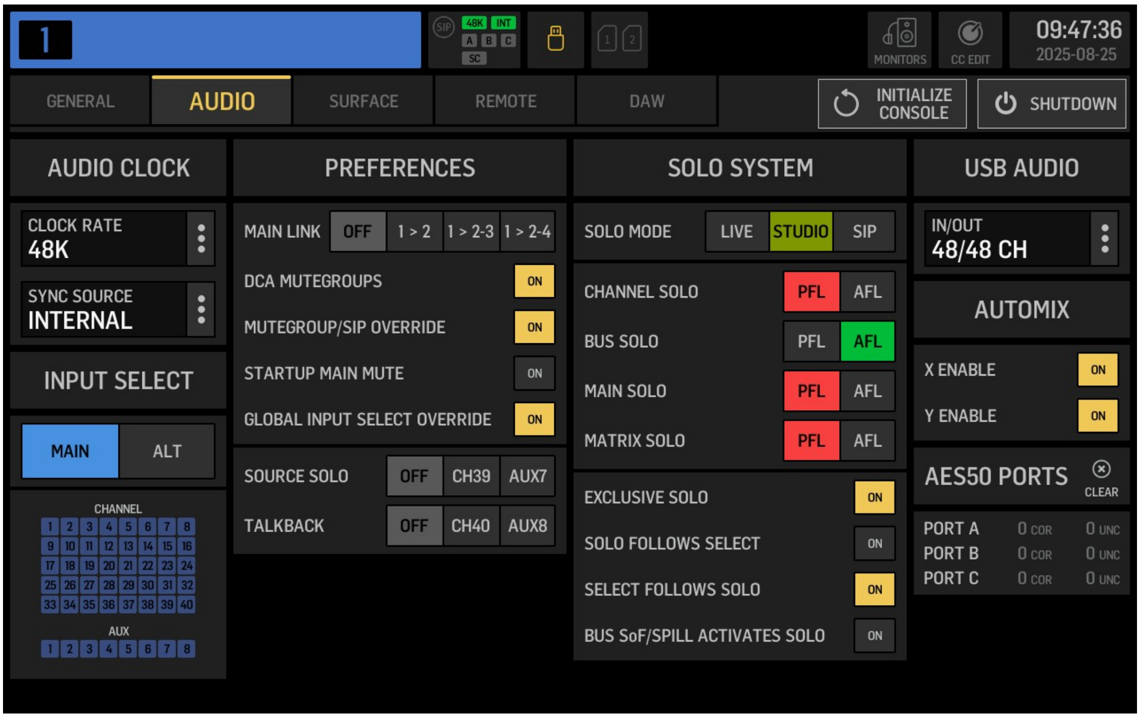

Figure 34: SETUP → AUDIO

AUDIO CLOCK controls the console's sample rate (Clock Rate) and the word clock source (Sync Source). Both the Clock Rate and the Sync Source are displayed at the top of the main display. If the console's clock is not synchronized, the Clock Rate and Sync Source icons at the top will blink red.

INPUT SELECT allows you to switch the MAIN and ALT inputs on all 40 Input channels and 8 Aux channels. Click on the channel grid to open the Input Select window. Clicking on individual channels on the left panel (MAIN/ALT INPUT) toggles between the MAIN (blue) and ALT (orange) inputs.

To toggle between the MAIN and ALT inputs on multiple channels at once, select the channels on the right panel (GLOBAL INPUT SELECT ENABLE) whose inputs should be controlled by the MAIN/ALT switch on the SETUP → AUDIO screen. Clicking on the MAIN/ALT switch toggles the inputs on all the selected channels. This switch can also be assigned to a CC button.

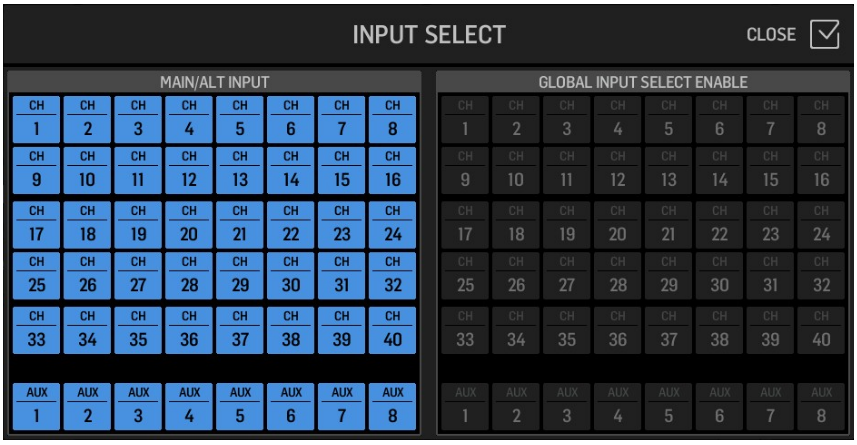

Figure 35: INPUT SELECT

PREFERENCES contains the following settings:

- MAIN LINK: links the levels of the MAIN1 to the other Mains so that the adjustments made to the MAIN1 level are also applied to the MAIN2; MAIN2 and MAIN3; or MAIN2, MAIN3 and MAIN4; like how a DCA controls the level of the channels assigned to it.
- DCA MUTEGROUPS: when on, each DCA also works as a Mute Group, so that when the DCA is muted, the channels assigned to the DCA are also muted.
- MUTEGROUP/SIP OVERRIDE: enables individual channels to be unmuted even if part of an active Mute Group, DCA Mute or if SIP (SOLO IN PLACE) is active.
- STARTUP MAIN MUTE: mutes all MAIN buses when the console is powered up.
- GLOBAL INPUT SELECT OVERRIDE: when activated, each channel's input can be manually changed from MAIN to/from ALT in the corresponding Channel Input section, even when the channel is selected in the GLOBAL INPUT SELECT ENABLE grid (Fig. 35).
- SOURCE SOLO: when active, a SOLO function for individual Sources in the ROUTING → SOURCES screen is available (Fig. 36). The SOLO signal is routed to channel 39 or AUX channel 7.
- TALKBACK: enables the Talkback function and selects the channel used to host the Talkback signal. See Chapter 7: MONITORS for details.

Figure 36: SOURCE SOLO

SOLO SYSTEM includes the following settings:

- SOLO MODE
  - LIVE: Each channel has an independent setting in the Channel Home screen (see Fig.

21) that determines whether the channel is heard on the Phones or Speakers outputs when soloed (see Fig. 37). The Solo Bus can be manually changed on the MONITORS screen (see Fig. 72) after activating Solo, overriding the preselection on each channel. All Mains, Buses, and Matrices remain unaffected when Solo is active.

- STUDIO: Each channel is routed to the Solo Bus currently active on the MONITORS screen. As with Live mode, all Mains, Buses, and Matrices remain unaffected when Solo is active (see Fig. 38).

- SIP (Solo In Place): When Solo is activated on a channel, all other input and AUX channels are muted. This affects what is heard on the Buses, Mains, and Matrices. When SIP is active, the SIP icon is displayed on the upper border of the screen. Additionally, a Solo Safe option (see Fig. 39) is available on each channel to prevent it from being muted when Solo is activated on another channel.

Figure 37: Live Solo Mode

Figure 38: Studio Solo Mode

Figure 39: SIP Solo Mode

- The Solo function can be set to be pre-fader (pre-fader listen, PFL) or post-fader (after-fader listen, AFL) on Channels, Buses, Mains and Matrices.

- EXCLUSIVE SOLO: Solo can only be active on one channel at a time.

- SOLO FOLLOWS SELECT: after activating solo, selecting a different channel also activates solo on the newly selected channel.

- SELECT FOLLOWS SOLO: activating solo on a channel also selects it.

- BUS SoF/SPILL ACTIVATES SOLO: when Spill mode or Sends On Faders mode is active on a

Bus, Main or Matrix, solo is activated on the selected Bus, Main or Matrix.

USB AUDIO sets the number of available input and output channels when connecting the console to a computer via USB 2.0 from 2x2 up to 48x48 channels.

AUTOMIX has independent switches to enable the X and Y groups for Automix processing. Please refer to Section 6.1: Input Channels for details.

AES50 PORTS provides a counter for the corrected (COR) and uncorrected (UNC) data transmission errors over AES50 to help identify faulty cables or connections. Click on RESET to restart the error counters.

SURFACE

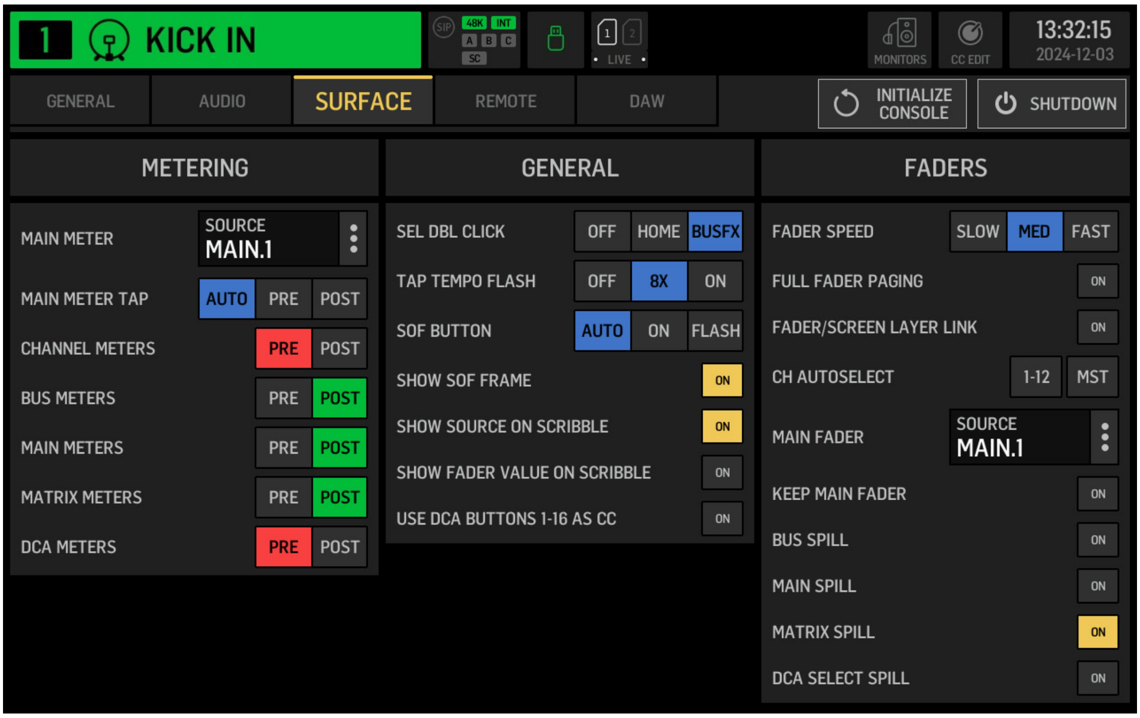

Figure 40: SETUP → SURFACE

This screen is organized into three sections. **METERING** contains the following options:

- MAIN METER SOURCE defines the bus or channel whose level is displayed on the LED meter next to the main display. You can choose between all the Main buses, Matrixes, Phones, Speakers or the selected channel.
- MAIN METER TAP:
  - AUTO: adjusts the Main Meter to pre- or post-fader, depending on which type of bus or channel is selected as Main Meter Source and its corresponding settings below in the screen.
  - PRE: the Main Meter level is pre-fader
  - POST: the Main Meter level is post-fader
- CHANNEL/BUS/MAIN/MATRIX/DCA METERS: the individual LED meters on each channel can be set to pre- or post-fader.

GENERAL contains the following options:

- SEL DBL CLICK determines what happens when pressing a SELECT button twice:
  - OFF
  - HOME opens the Channel Home screen.
  - BUSFX opens the INS1 (Insert Point 1) screen on Buses, Mains and Matrices (or INS2 if INS1 is unused). Input channels and Aux channels behave just like when the HOME option is active.
- TAP TEMPO FLASH: determines the blinking behavior of the TIME parameters of delay effects when assigned to a CC button for Tap Tempo. Note that this setting does not apply to the blank button on the right side of the main display, as this button always blinks when it can be used for Tap Tempo.
  - OFF: the CC button does not blink.
  - 8X: the CC button blinks 8 times each time the TIME parameter is adjusted and then turns off.
  - ON: the CC button always blinks.
- SOF BUTTON:
  - AUTO: SOF FLIP button blinks when Alternative SOF Mode is on and stays lit when the Standard SOF Mode is on
  - ON: SOF FLIP button remains lit when either Alternative or Standard SOF Mode is on
  - FLASH: SOF FLIP button blinks when either Alternative or Standard SOF Mode is on
- SHOW SOF FRAME: a yellow frame on the top left corner of the main display indicates that SOF mode is on (Fig. 41).
- SHOW SOURCE ON SCRIBBLE: the Source assigned to each channel is shown on the top of each scribble strip. When inactive, the channel mono, stereo or mid/side configuration is shown instead of the Source.
- SHOW FADER VALUE ON SCRIBBLE: the channel's fader value in decibels is temporarily shown on the scribble strip instead of the channel icon when adjusting the fader.
- USE F1-F3 AS CUSTOM CONTROLS mirrors the Custom Control layers 1-3 on the 4-Channel Section. F1 opens the 4 knobs and top row of 4 buttons of the first layer. F2 opens the second layer, and so on. Please refer to Section 4.8 for details on how to assign and operate Custom Controls.
- SCREEN FOLLOWS CH STRIP: the main display automatically shows the screen that corresponds to the processing block that is currently being edited in the Channel Strip Section.
- CH STRIP TOUCH SELECT: the knobs on the Channel Strip Section are touch sensitive. Upon touching a knob, the corresponding processing block is selected, even without turning the knob.
- USE DCA BUTTONS 1-16 AS CC (WING COMPACT): allows using the 16 quick access buttons on the DCA layer as Custom Controls.

Figure 41: SOF Frame

FADERS contains the following options:

- FADER SPEED adjusts how quickly the motorized faders move when changing from one Scene to another, changing layers, paging through layers, etc. Note that this setting has no influence on the audio.
- ALTERNATIVE SOF MODE: the destination Bus, Main or Matrix of the SOF mode remains fixed even when another channel is selected. To enter the Alternative SOF Mode, hold down the SELECT button on the desired destination channel. To exit, press the SOF FLIP button. Please refer to Section 4.9 for details.
- RIGHT SECTION SENDS ON FADERS (WING): when active, the right fader section on WING can also be used in SOF MODE to control the sends to Mains, Matrices and Buses. If disabled, the right fader section can only be used in SOF mode to select the destination bus.
- FULL FADER PAGING: the fader layer navigation buttons (←4 and 4→) change the entire fader “pages” instead of scrolling 4 channels at a time.
- FADER/SCREEN LAYER LINK: when disabled, the main display and fader layer can show different channels
- CH AUTOSELECT: automatically changes the channel selection to the channel whose fader is adjusted. These settings can be activated for each fader section independently.
- MAIN FADER (COMPACT): selects the channel controlled by the main fader
- KEEP MAIN FADER (WING COMPACT): when active, the main fader stays assigned to the same channel, regardless of what layer is selected in the main fader section.
- DCA SPILL ON SELECT (WING COMPACT): when active, selecting a DCA activates SPILL mode for that DCA.
- USER LAYER LINK (WING): the User Layers in each fader section (Left, Center, Right) can be linked so that activating one User Layer in a fader section also activates the corresponding User Layer on another fader section.
- DCA SPILL (WING): when active, after selecting a DCA, only the channels currently assigned to it are displayed on the left fader section.
- BUS/MAIN/MATRIX SPILL (WING COMPACT): when active, Spill mode is activated automatically when selecting a Bus, Main or Matrix with the 16 buttons in the Main Fader section
- DCA SELECT SPILL (WING COMPACT): when active, pressing the SELECT button on a DCA activates Spill mode

REMOTE

MIDI REMOTE CONTROL determines if the remote control of multiple console parameters via MIDI is off or done via the DIN5 or USB connections. Details about which MIDI Channel, MIDI CC (Continuous Controller) and MIDI Note controls which console parameter can be seen by clicking on the corresponding information icon.

Please refer to the Section 5.6 LIBRARY for details on how to create and run a SHOW file.

ENABLE HA REMOTE determines whether other consoles connected to the corresponding AES50 ports can control WING's onboard preamps. **ENABLE CUST SYNC** determines whether the customization information from other consoles can be received over AES50.

After enabling the general preamp remote control (ENABLE HA REMOTE) and customization sync (ENABLE CUST SYNC) on this screen, HA REMOTE and CUST SYNC for individual Sources must be enabled/disabled in the ROUTING → SOURCES page (Fig. 31) (see Section 5.4).

While on the SOURCES page, select a Sources and click on the AES50 port for which HA REMOTE should be enabled. Hold down the desired AES50 button to enable remote control from this port for all the channels in the same Source Group.

RCV CUST allows the console to receive the customization information for the selected Source from another device, even when HA REMOTE is not enabled. Note that the Sources must be routed from one console or stage box to another. The general and individual HA REMOTE and CUST SYNC switches must also be enabled to allow their respective communications. If any of these steps are missing, the buttons light up dark yellow. When the corresponding function is active, the button lights up bright yellow.

The CLEAR button on the top right of the SETUP → REMOTE → HA REMOTE section clears the individual REMOTE switches for all AES50 Sources in the ROUTING → SOURCES screen.

NETWORK allows you to select either DHCP or Static IP Modes for communication with the following remote-control apps:

- WING Edit: desktop application for Mac, Windows, Linux and Raspberry Pi
- WING Copilot: mobile application for Android and iOS
- WING Q: mobile application for Android and iOS, focused on Bus and Matrix control for personal monitoring uses.

For mobile devices, desktop computers and the console to communicate with each other and enable remote control, they must all be connected to the same network. The console must always be connected to the switch/router using an Ethernet cable. The computer can be connected either wirelessly or with an Ethernet cable. Mobile devices will most likely be connected wirelessly. Up to 16 devices can remote-control WING.

REMOTE LOCK allows you to block the remote control over IP/OSC port 2223 and IP/TPC port 2222.

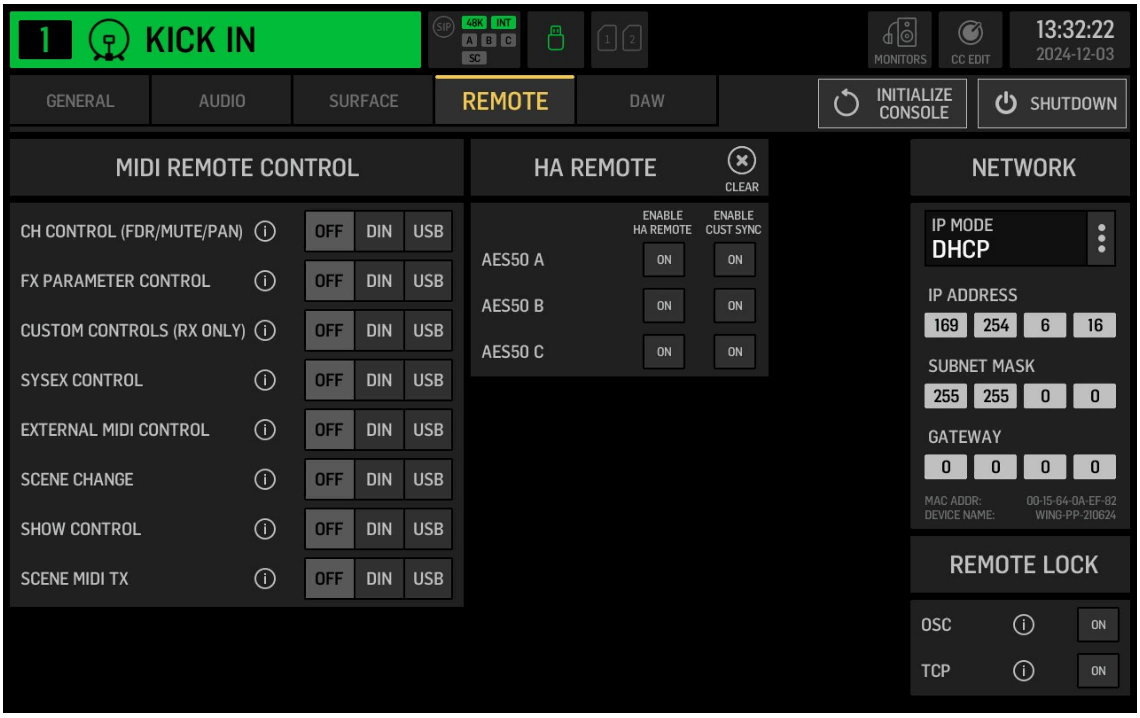

Figure 42: SETUP → REMOTE

DAW

WING allows DAW control through the CCs, including transport control, faders and specific parameters.

- DAW ENABLE: allows communication between the console and the DAW. After enabling this setting, press the DAW button (WING) in the Custom Controls section (Fig. 20) to switch the faders to DAW control. On WING COMPACT and RACK, the DAW button must be assigned to a CC.
- CONNECTION: select between DIN5 or USB MIDI connection with the DAW
- EMULATION: select between MCU or HUI protocols
- USE UPPER CC FOR DAW (WING): allows using the top part of the Custom Controls section for DAW control instead of for CC functions
- DISABLE WHEEL DURING PLAY: when the DAW is playing back, navigation with the wheel is disabled.
- CONFIGURATION: determines the number of DAW controllers emulated by the console and shows the control surfaces on the console that correspond to each emulated controller.
- PRESET: multiple DAW presets are available.
- LOAD: loads the selected DAW preset

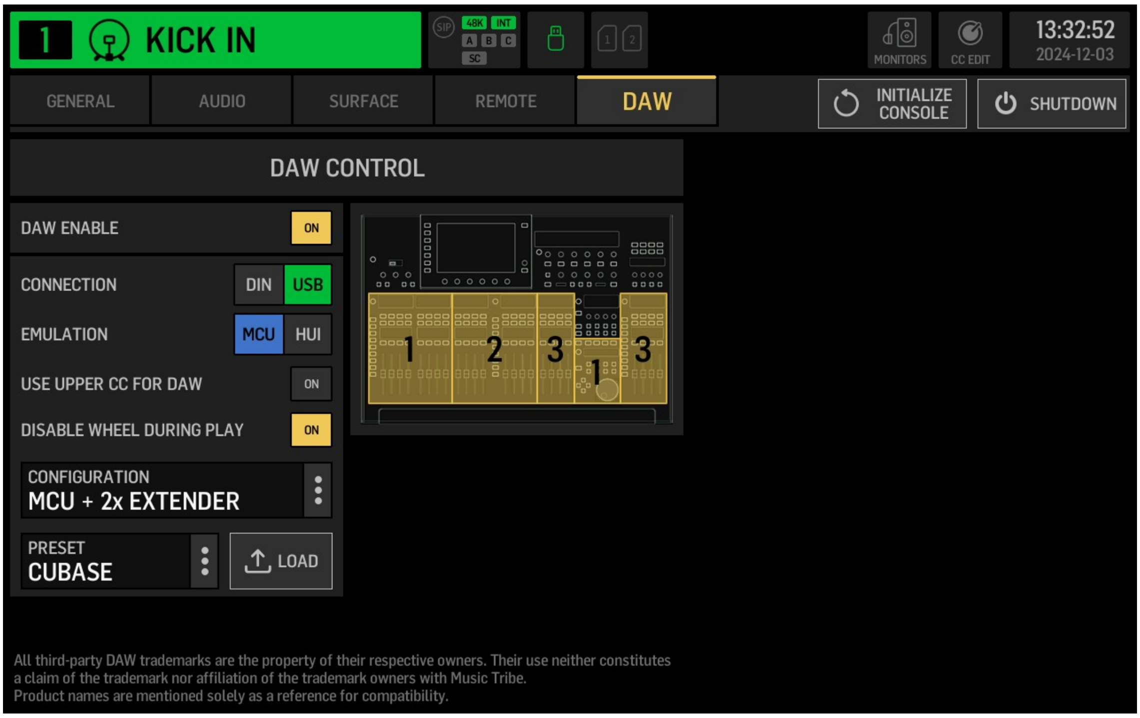

Figure 43: SETUP → DAW

5.6 LIBRARY

The LIBRARY section (Fig. 44) manages the saving and recalling of different types of presets and audio clips. This section also manages outgoing MIDI messages when recalling specific files, as well as recalling files with incoming MIDI messages.

The screen is divided into two sections: the left panel shows the files saved in the console's internal storage, on the USB stick or within a Show file. The right panel has dedicated tabs for each file type that can be managed in the Library. Each tab has filetype-specific properties.

The files on the left panel are displayed normally or greyed out, depending on which tab is open on the right. For example, when the SNAP tab is open, Snapshots are displayed normally in the left panel. All other filetypes are greyed out.

It is possible to load a Show, Snapshot or Snippet automatically on startup by renaming the desired file and including "STARTUP" at the beginning of the file name, for example "STARTUP_myshow.show", "STARTUP-mysnap.snap", "STARTUP.snip", etc. These files must be placed in a folder named "STARTUP" in the root directory.

SHOW

Show files allow Scene navigation and recall with the dedicated hardware and software buttons, as well as with Custom Controls and MIDI data. After a Show has been created or opened, all files that can be added to the Show are displayed normally (not greyed out) in the left panel. Any filetype (Snapshot, Snippet, Clip or FX preset) added to a Show is referred to as a Scene.

To add a file to a Show, select it on the left panel and click on ADD on the top of the screen or on ADD ITEM TO SHOW in the SHOW tab. When a Show is open and shown in the left panel (Fig. 45), new Snapshots, Snippets, effects and channel presets can also be saved and added automatically to the Show with the SAVE + ADD button found in the corresponding SNAP, SNIP, FX or CHAN tab.

The following options are available in the SHOW tab:

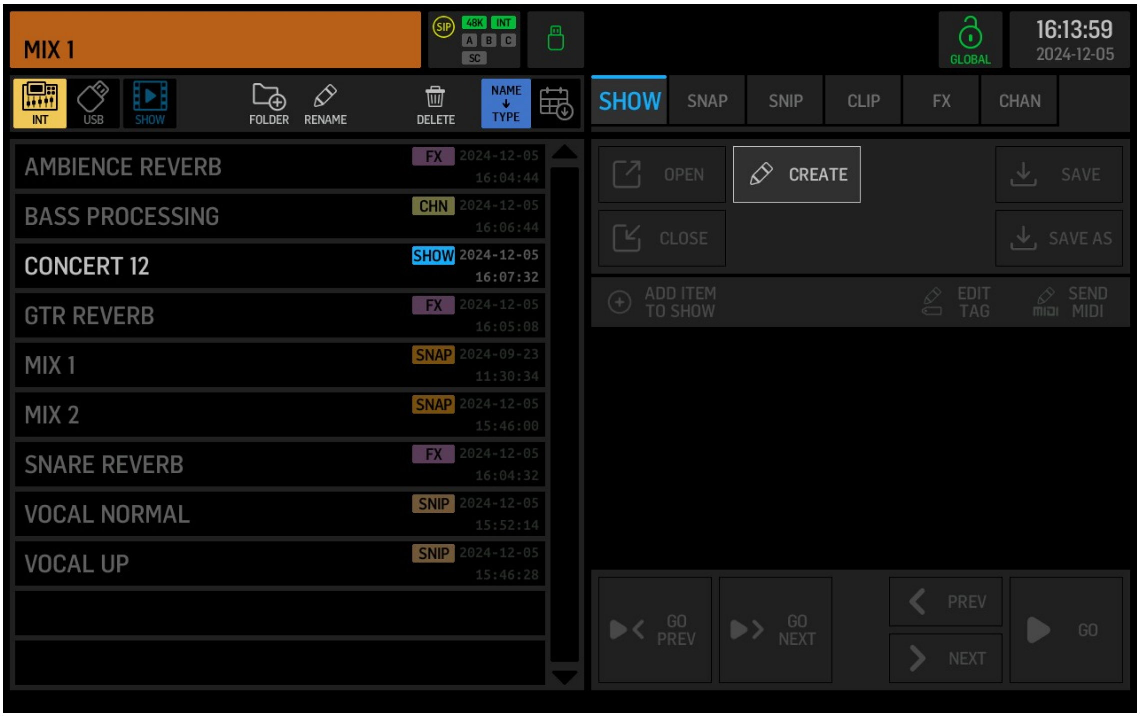

Figure 44: LIBRARY

- RENAME: Scenes within a Show can be renamed without altering the original file name. If renamed, the original Scene name and filetype are also displayed.
- SKIP: the selected Scene is temporarily disabled and ignored by the navigation and recall controls.
- LINK: links a Scene with the next one. When recalling the first Scene, the next one is also loaded.
- UP/DOWN: moves the selected Scene up or down the list. If a Scene is linked and moved, the Scene remains linked to the new Scene immediately below it, not to the one it was originally linked to.
- EDIT TAG: assign a tag from #1 to #128 (including the # sign) to a Scene to then recall it with a Custom Control (CUSTOM CONTROLS → FUNCTION: SCENE RECALL → SCENE TAG: SCENE #) or through MIDI. For example, tag #1 is assigned to a Scene in a SHOW. The Scene tag #1 is assigned to a CUSTOM CONTROL button. When the SHOW is open and the CUSTOM CONTROL is pressed or the corresponding MIDI message is received, the Scene is recalled.
- SEND MIDI: specify the MIDI TX STRING sent via the DIN5 or USB MIDI connection when the Scene is recalled. Use hexadecimal notation, separated by a blank space or a comma, and use “D:” or “U:” to specify if the MIDI TX STRING should only be sent via DIN5 or USB. For example: C003,B12250,U:B07E7F
- GO PREV: the previous scene is loaded
- GO NEXT: the next scene is loaded
- PREV: select the previous Scene
- NEXT: select the next Scene
- GO: load the selected Scene

Figure 45: LIBRARY → SHOW

SNAP

Snapshots (Fig. 46) always store all the console's parameters but can selectively load only some of them, as defined by the Snapshot's Scope.

Some cells include dropdown menus that let you select individual channels within a group. If all channels in the group are selected, the cell appears fully lit. If only part of the group is selected, the cell appears dimly lit.

- SAVE: creates a new Snapshot with all the current console configurations
- SAVE + SCOPE: creates a new Snapshot with all the current console configurations but also allows you to specify which parameters will be loaded when the Snapshot is recalled (Scope).
- UPDATE: the changes in the parameters within the Scope of the selected Snapshot are saved.
- PARTIAL UPDATE: a Scope window opens. The changes in the parameters selected in this window are saved.
- EDIT SCOPE: change the parameters that are loaded when recalling the selected Scene.
- LOAD: the selected Scene is recalled, according to the Snapshot's Scope.
- PARTIAL LOAD: a Scope window opens. Only the parameters selected in this window are recalled. If the NO GLOB button is active, the GLOBAL SAFE is ignored and the corresponding parameters are overwritten, if part of the scope that is being loaded.
- RESTORE: recalls the selected Snapshot completely, regardless of its Scope, including global SAFES.

The **Scope** (Fig. 47) of a Snapshot consists of the following categories:

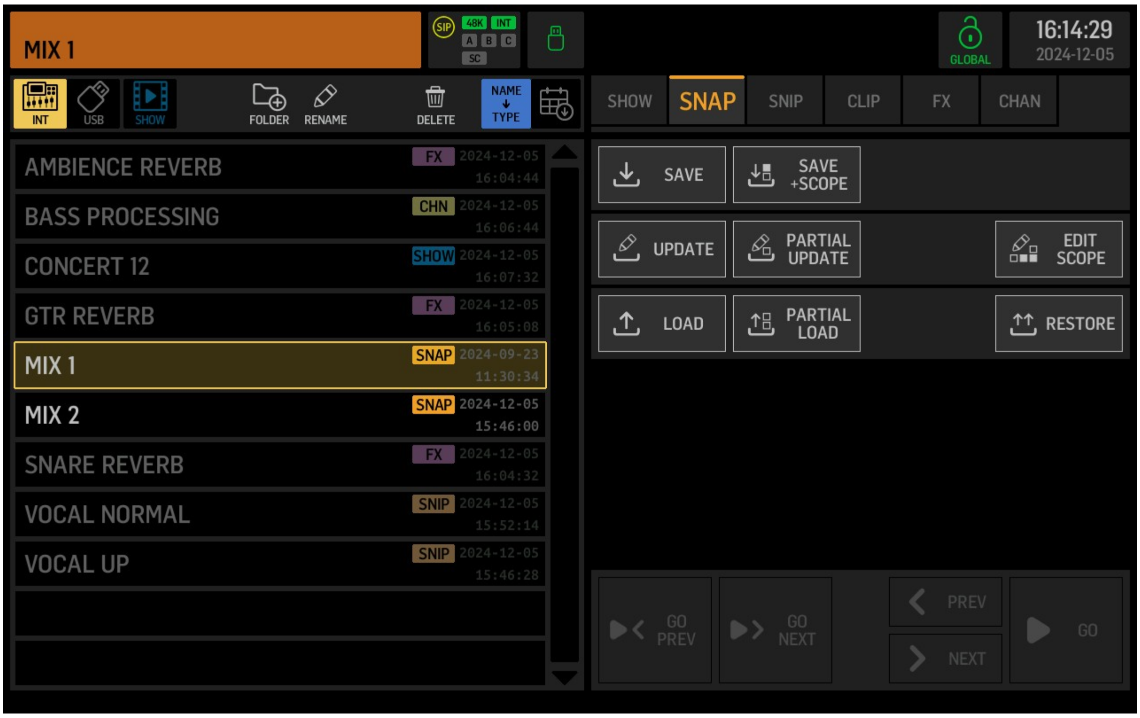

Figure 46: LIBRARY → SNAP

- INPUT CHANNELS 1-40
- AUX CHANNELS 1-8
- BUS 1-16
- MAIN 1-4
- MATRIX 1-8
- All SOURCE GROUPS (click on each group to select individual Sources)
- All OUTPUT GROUPS (click on each group to select individual Outputs)
- DCA 1-16
- MUTE GROUP 1-8
- FX slots 1-16
- CONFIGURATION
  - CONFIG (console settings)
  - SFC (multiple Surface settings including lights, RTA settings, User Layer Link, Spill modes, etc.)
  - PREFS (preferences)
  - L (WING left fader section)
  - C (WING center fader section)

- R (WING right fader section)
- CC (CUSTOM CONTROLS) layers:
  * GPIO
  * USER and DAW 1-4 (WING)
  * CC layers 1-16 (WING and WING RACK)
  * USER 1-4 (WING COMPACT)
  * MAIN/MATRIX (WING COMPACT)
  * DCA 1-4 (WING COMPACT)
- COMPACT (WING COMPACT fader layers)
  * CH 1-12
  * CH 13-24
  * CH 25-36
  * CH 37-AUX 8
  * BUSES
  * MAIN/MATRIX
  * DCA
  * USER 1
  * USER 2
- RACK (WING RACK front panel layers)
  * CHANNELS (Input Channels 1-40)
  * AUX (Aux Channels 1-8)
  * BUSES (Buses 1-16)
  * MAIN/MTX (Mains 1-4 and Matrices 1-8)
  * DCA (DCA 1-16)
- EXTERNAL
- VIRTUAL (8 virtual layers)

· CONTENTS
  - CUST (Source customization)
  - TAGS

- CONN (Source assignment and MAIN/ALT selection)
- IN (Gain, Trim, Phantom Power, Phase Invert)
- FILTER
- DELAY
- GATE
- DYN (Dynamics)
- PRE (pre-fader insert point)
- POST (post-fader insert point)
- EQ
- PAN (channel panning and width)
- MAIN (Main Sends 1-4, individually selectable)
- SEND (Bus 1-16 and Matrix 1-8 sends, individually selectable)
- FDR (channel fader level)
- MUTE
- CONFIG (solo options, TAP width and position, processing sequence)

Figure 47: SCOPE

Clicking on the **GLOBAL** button (Fig. 48) allows you to select channels, parameters or settings that will remain unchanged even when recalling other Scenes. When one or more items are selected, the GLOBAL icon turns red.

Figure 48: GLOBAL ICON

SNIP

Snippets (Fig. 49) are used to store specific parameters instead of all the console settings. To create a Snippet, click on REC, modify the desired parameters, click on STOP and SAVE the Snippet. The number of modified parameters is shown on the screen.

REC FOCUS allows you to limit the parameters and channels which are stored in the Snippet. Clicking on the REC FOCUS button opens a screen with the same parameters as a Scope. Channels or contents outside of the REC FOCUS' Scope are not saved within the Snippet. If the Scope has been edited, the REC FOCUS button turns light blue.

Parameters can also be added to a Snippet without manually adjusting them. Click on ADD ITEMS and select the desired contents from the Scope window. Likewise, items can be removed from a Snippet.

CLEAR ALL removes the parameters that have been edited.

UPDATE is used to update the parameters already stored within an existing Snippet.

PARTIAL LOAD is used to recall only specific contents from a Snippet.

LOAD FOCUS sets the REC FOCUS to the parameters already contained in the selected Snippet.

LOAD recalls all the parameters in a Snippet.

Figure 49: LIBRARY → SNIP

CLIP

The CLIP tab (Fig. 50) allows you to play back 2 or 4-track 44.1/48 kHz and 16/24-bit WAV files. The files created with the 4-track USB recorder are also available here. The channels from the CLIP/USB player must be assigned to one or more channels on the ROUTING screen to be heard. Please refer to Chapter 8 for details.

Figure 50: LIBRARY → CLIP

FX

It is possible to store the settings of any of the 16 FX slots through the FX tab (Fig. 51). Select the desired FX slot from the dropdown menu and click on SAVE. When an existing FX preset is selected, the effect saved in it is also displayed.

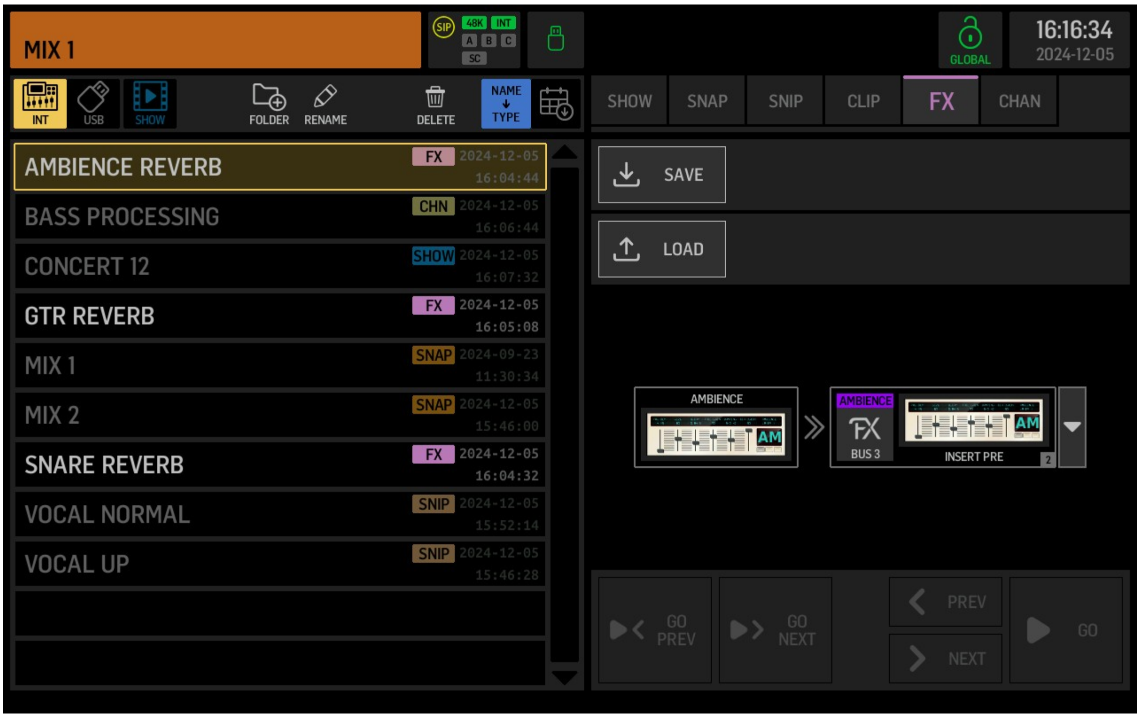

Figure 51: LIBRARY → FX

CHAN

Any channel can also be saved as a preset (Fig. 52). The channel currently selected whose settings are to be saved or to whom the saved settings are to be applied is shown on the right.

The CHAN preset's Scope can also be updated with the UPDATE SCOPE button.

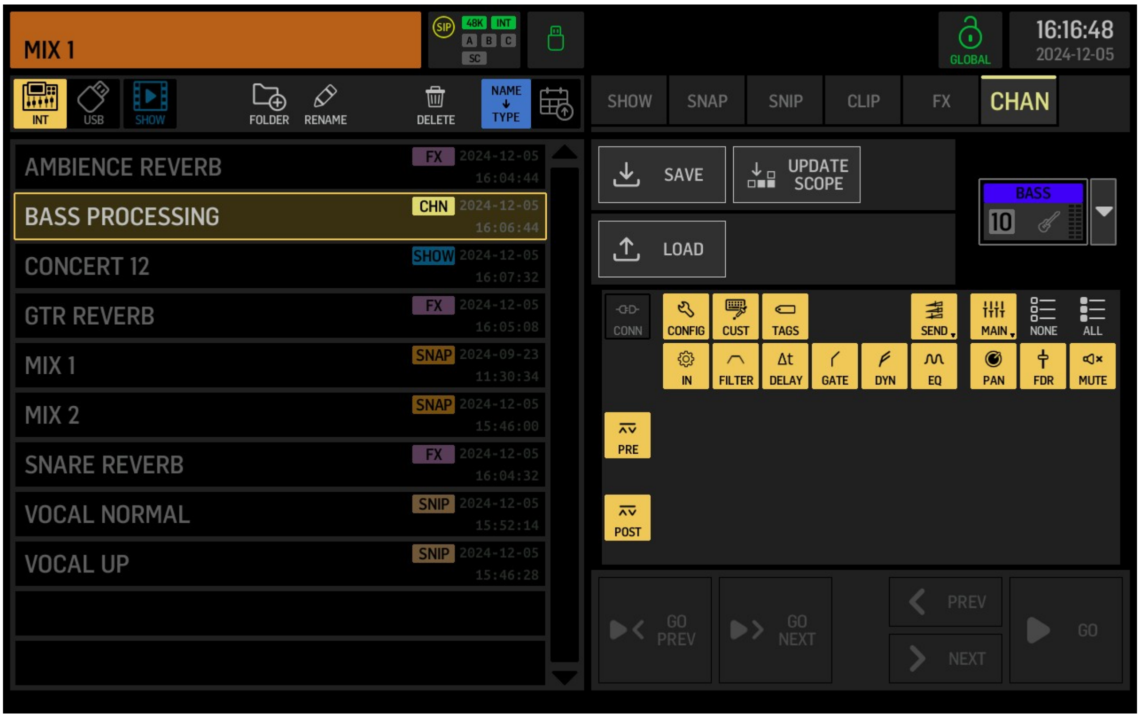

Figure 52: LIBRARY → CHAN

5.7 OVERVIEW

Press the VIEW button in a fader section to open the OVERVIEW screen (Fig. 53) with channels corresponding to the current fader layer.

It is possible to select different fader layers with the blue-highlighted buttons on the right side of the screen (except when on the WING left fader section OVERVIEW, Fig. 54). When doing so, the hardware control surface changes accordingly. It is also possible to switch to different fader sections with the green-highlighted buttons on the bottom right corner of the screen.

Note that WING RACK has a hardware (RACK) and a virtual fader section. The RACK section consists of four channels shown on the screen and on the 4-channel section on the front panel. The virtual section consists of 8 channels and operates independently of the hardware 4-channel section.

WING RACK and COMPACT also have additional virtual fader layers (Fig. 55) accessible by clicking on the empty buttons on the right of the screen. Rename and populate these layers on the CONFIG screen (see Section 5.10).

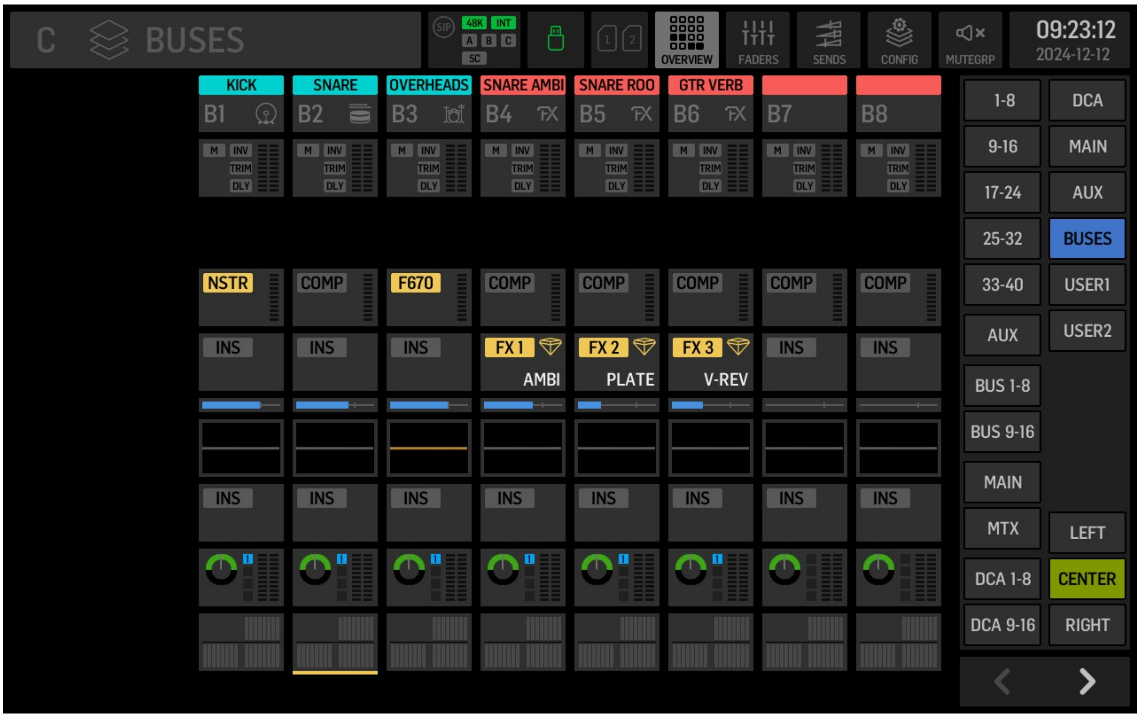

Figure 53: C fader section OVERVIEW screen (WING)

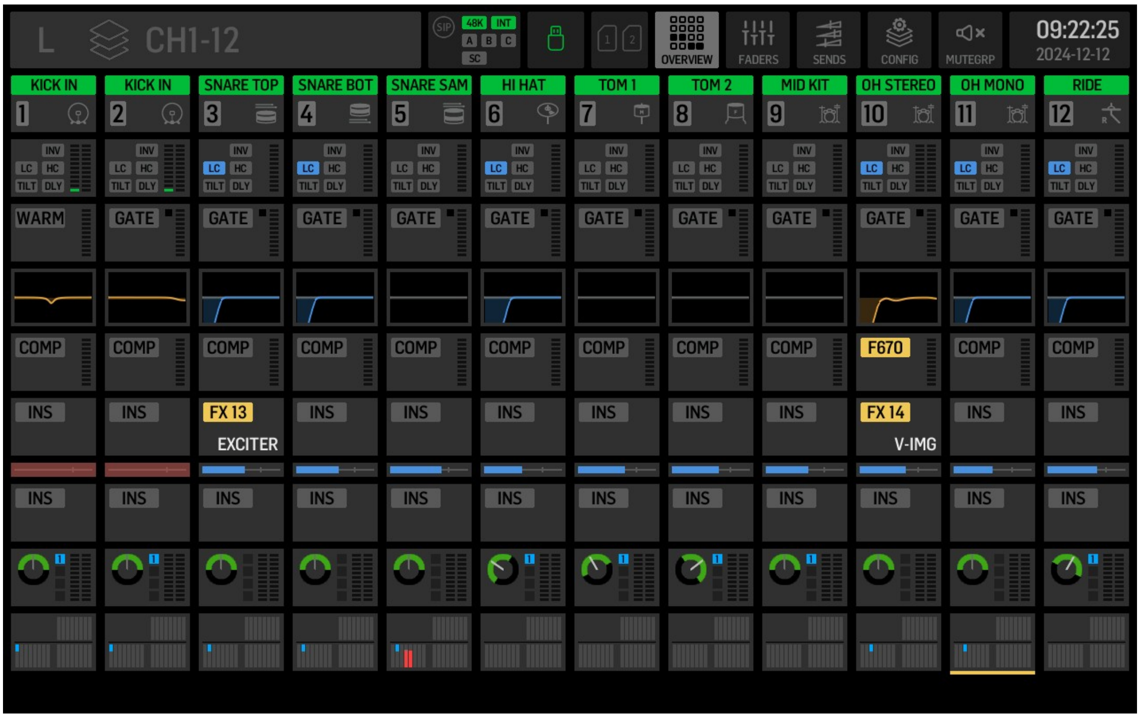

Figure 54: L fader section OVERVIEW screen (WING)

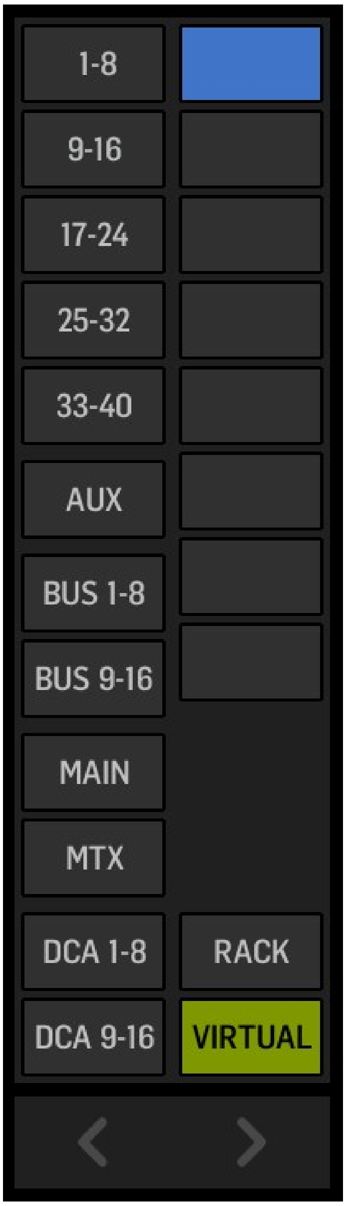

Figure 55: Additional virtual fader layers on WING RACK

5.8 FADERS

Open the FADERS screen (Fig. 56) by clicking on the FADERS button on the top of the screen after a VIEW button on a fader bank has been pressed. A virtual fader is available for each channel. Just like the OVERVIEW screen, the FADERS screens for the center and right section also allow selecting different fader layers and switching to other fader sections.

The first horizontal meter shows the gain reduction applied by the Gate slot. The square indicator to the left of the meter lights green when the gate is open. If the Gate processor is swapped for a processor other than a noise gate, the meter turns yellow, and the square indicator disappears. Other metering variations appear depending on the effects being used.

The second horizontal meter shows the gain reduction applied by the Comp slot. When using the AUTO RIDER processor, the meter turns green, and a square indicator shows when the processor engages.

To the right of the fader, a meter shows the channel level. The assignments to the 16 DCAs, 4

MAINS and 8 MUTE GROUPS are shown.

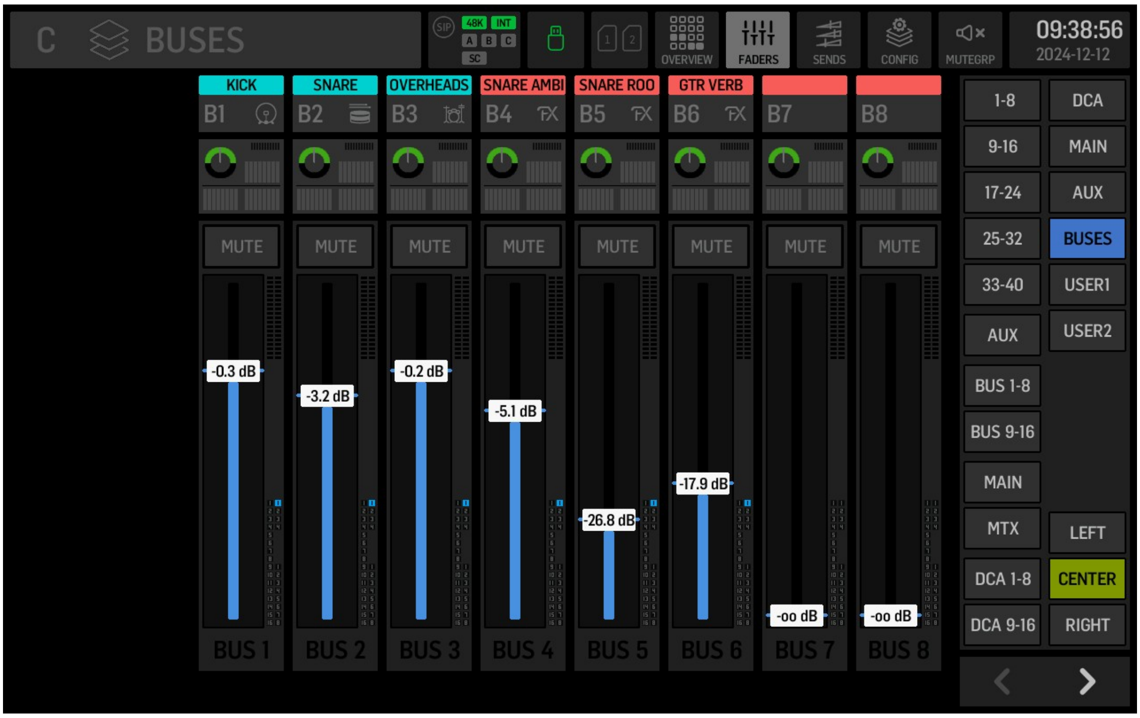

Figure 56: C fader section FADERS screen

5.9 SENDS

The SENDS screen (Fig. 57) can be understood as a "virtual Sends On Faders mode". To access the SENDS screen, click on the SENDS icon to the right of the OVERVIEW and FADERS buttons.

The top left panel shows the available destination layers:

- BUS 1-8
- BUS 9-16
- MAIN
- MATRIX

Once a layer has been selected, the individual destination BUS is chosen on the top right. Use the buttons on the right side of the screen to change the channel layers that can be sent to the selected destination.

5.10 CONFIG

Hold down the VIEW button on a fader section to open the CONFIG screen (Fig. 58). Alternatively, click on the CONFIG button next to the OVERVIEW, FADERS and SENDS buttons.

This screen allows you to edit all the fader layers, including the default ones. Besides channel levels, it is also possible to assign Send levels, effect parameters and MIDI CCs.

The layer to be edited is selected on the right side of the screen. Use the arrows on the bottom right of the screen to access additional pages in most of the fader layers not accessible by default.

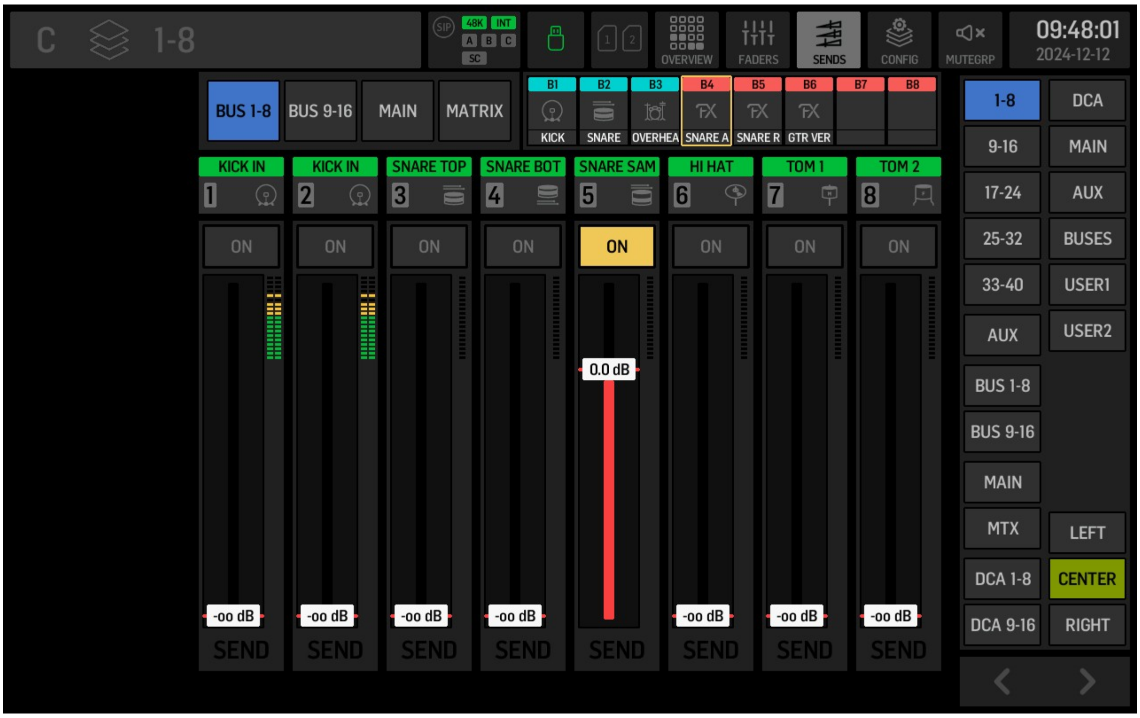

Figure 57: C fader section SENDS screen

The assignable channels, parameters, Send levels and MIDI CCs are selected at the top of the screen. The following are available:

- CHANNELS: Input channels 1-40 and Aux channels 1-8
- BUS/MAIN: Bus 1-16, Main 1-4, Matrix 1-8
- DCA: DCA 1-16
- SENDS: sends from the 40 Input channels and 8 Aux channels to Bus 1-16, Matrix 1-8 and Main 1-4
- BUS → BUS: sends from Bus 1-16 to BUS 1-16, Matrix 1-8 and Main 1-4
- FX: select an effect slot (1-16) to assign a parameter of the selected effect to the fader layer
- MIDI CC: MIDI Continuous Controllers 0-127 for channels 1-16

To assign a channel, parameter, Send level or MIDI CC to the desired fader layer, drag and drop from the top grid to the bottom grid. To assign multiple items at once, click on all the desired items and then drag the items to the fader layer.

To remove a fader from the layer, drag the desired fader outside of the bottom grid.

Fader assignments can be rearranged by dragging.

Layers can be reset or renamed with the CLEAR and RENAME buttons, respectively.

When assigning a Send to a Bus, Matrix or Main, the white box with an S is the selected channel send level to the Bus (Fig. 59).

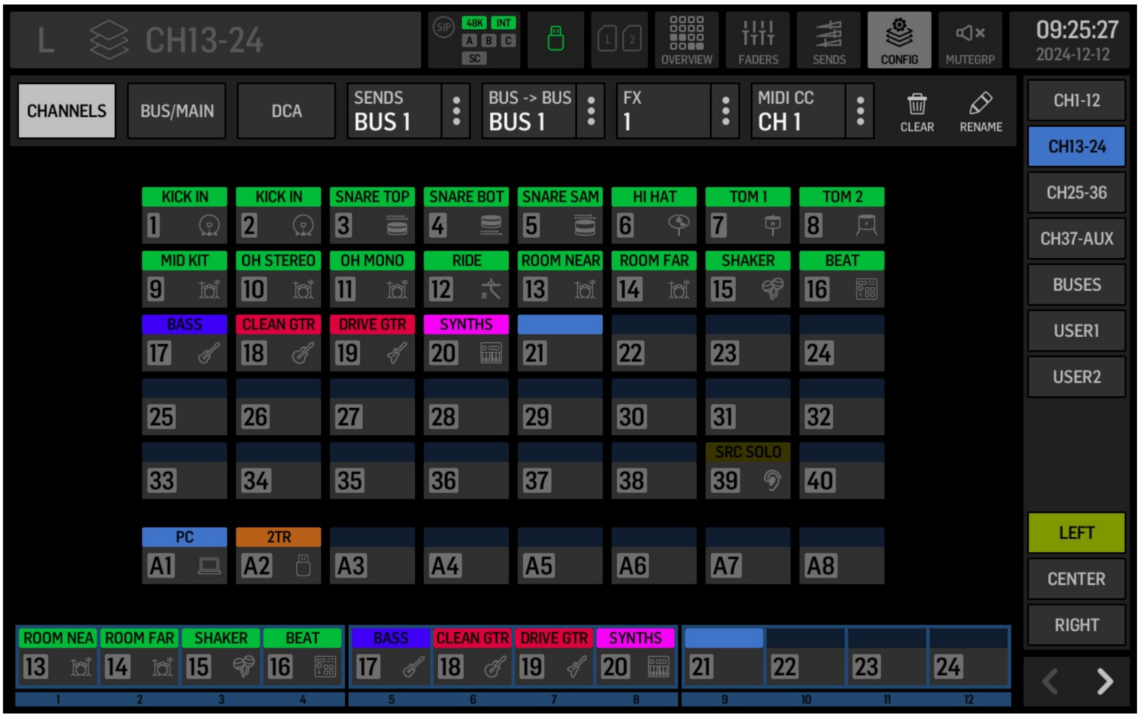

Figure 58: CONFIG screen

Figure 59: Selected channel Send to Bus/Matrix/Main

When assigning a Send level, it is possible to use the Source or Bus customization (icon, color and name), depending on which button is active when assigning the Send level to the fader layer (Fig. 60). Each layer can be renamed.

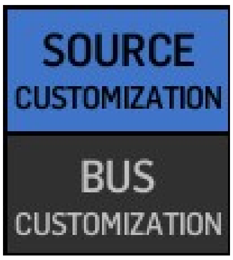

Figure 60: Source/Bus Customization

5.11 MUTE GROUPS

The MUTEGRP button next to the CONFIG button opens the Mute Groups screen. An overview of the 8 Mute Groups and the channels, Buses and Sources assigned to each, is shown.

To assign a channel to a Mute Group, click on the grid corresponding to the desired Mute Group and then select the channels, Buses, Mains, Matrices and Sources.

The 8 Mute Groups can be activated by clicking on the buttons to the right of the screen. To rename a Mute Group, click on the top part of each Mute Group. Note a Mute Group never overrides individual channel mutes. Clicking on the UNASSIGN button clears up all the assignments of the selected Mute Group.

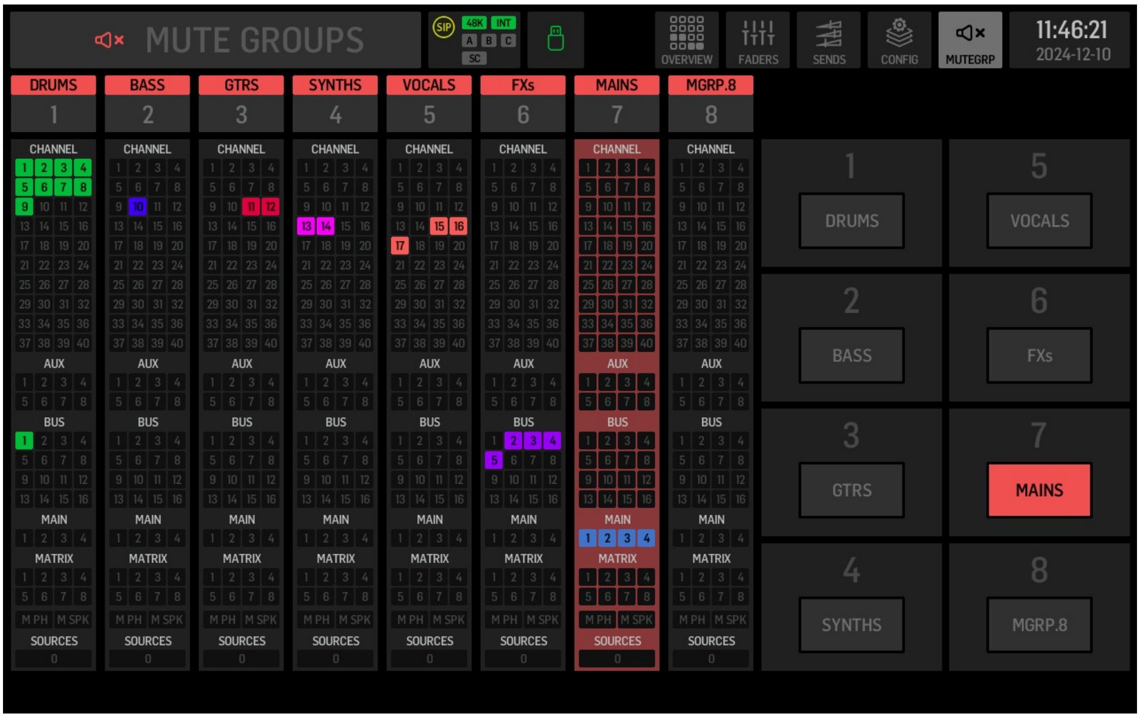

Figure 61: MUTE GROUPS screen

Section 6

Channel Processing

Besides the Input and Aux channels, all Buses, Matrices and Mains also have a dedicated channel with its respective processing capabilities. Effects like reverbs and delays can be loaded as insert points in any Bus, instead of being set up through dedicated effect buses or returns. Input channels have the most flexible processing options. Aux channels offer reduced processing options compared to Input channels. Bus, Main and Matrix channels all have the same processing options (see the sections below for details).

6.1 INPUT Channels

The order of all processing slots and the TAP point can be adjusted by clicking on the wrench icon on the bottom-left corner of the Channel Home screen and dragging the blocks (Fig. 21). The GATE, EQ, COMP, and INS1 slots are always pre-fader and the INS2 slot is always post-fader.

The TAP point defines where a copy of the signal is derived for its use on User Signals or for the sends configured in TAP mode, for example.

The following TAP point positions are available:

1. INPUT: Located immediately after the preamp, before any processing.
2. FILTER: Positioned after the low-cut and high-cut filters.
3. TAP 3: Between the first and second processing slots, based on the user's configuration.
4. TAP 4: Between the second and third processing slots.
5. TAP 5: Between the third and fourth processing slots.
6. PRE FDR: Between the fourth processing slot and the channel fader.
7. POST FDR: Between the channel fader and the INS2 slot.
8. POST PROC: After the INS2 slot and before the WIDTH control.

CHANNEL INPUT / TRIM and BALANCE / FILTER / DELAY (POST)

This screen offers the following options (Fig. 62):

- CHANNEL INPUT: assign a Source to the channel's MAIN and ALT inputs and toggle between them.
- INPUT SELECT:

- INDIVIDUAL: makes toggling between the MAIN and ALT inputs possible only with the individual switches on the Channel Input section or on the Input Select screen (Fig. 62)
- GLOBAL: the channel's INPUT can be switched with the GLOBAL MAIN/ALT switch in the SETUP → AUDIO → INPUT SELECT section (Fig. 34).

• TRIM & BALANCE:
  - TRIM: digital level control up to ±18 dB gain
  - BALANCE: positive values attenuate the left channel and boost the right channel by the same amount up to 9 dB. Negative values have the opposite effect. If the SOURCE is mono, this control has no effect.
  - INVERT: phase inversion
  - LED meter (post TRIM)
  - FILTER: clicking on the name opens a screen with additional controls (Fig. 63)
    * TOOL FILTER: Tilt EQ, Maxer, All-Pass 90°, All-Pass 180°
    * LC SLOPE: 6, 12, 18 and 24 dB/octave
    * HC SLOPE: 6 and 12 dB/octave
    * LC FREQ: low cut-off frequency
    * HC FREQ: high cut-off frequency
    * Depending on the selected tool filter, different options are available

• DELAY (POST): post-fader delay

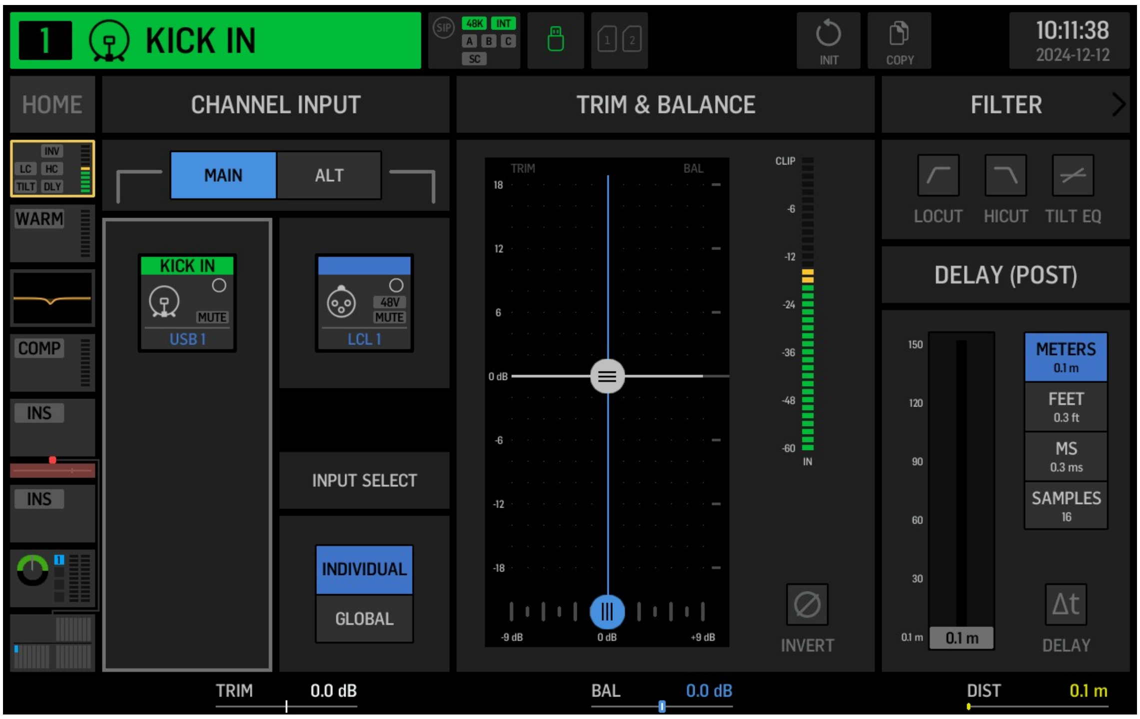

Figure 62: INPUT screen

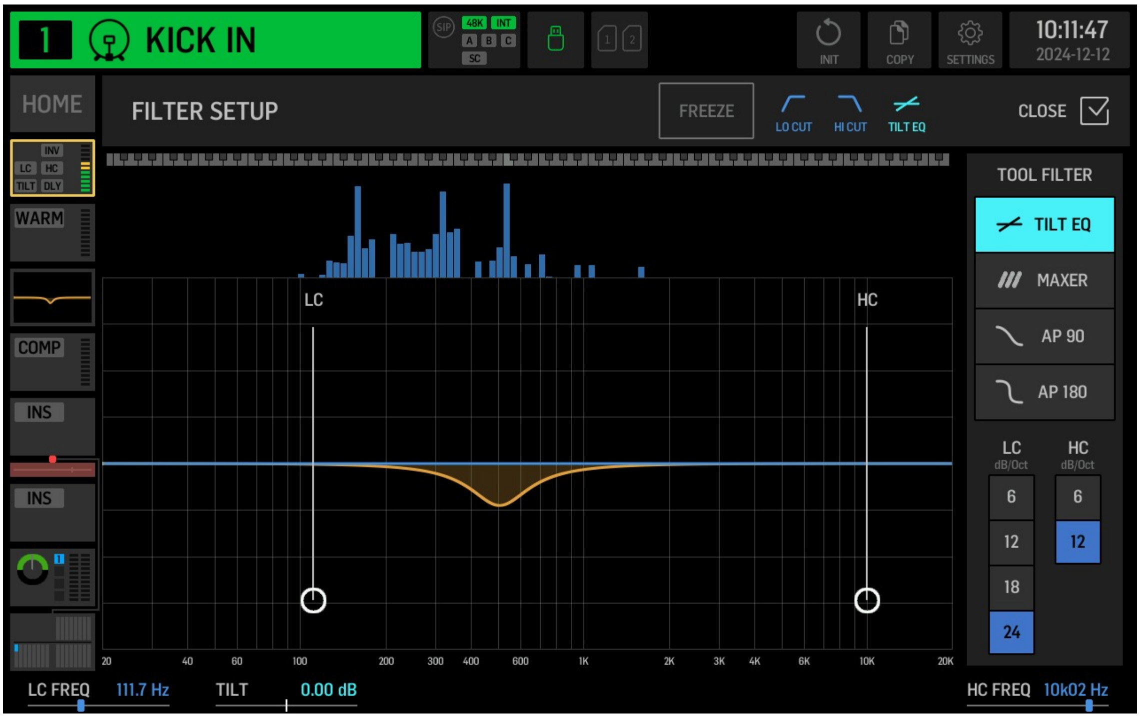

Figure 63: FILTER screen

GATE

The Gate processing slot shows different parameters, depending on the processor model chosen in the dropdown menu on the top of the screen. By default, a Gate/Expander is selected. The same dynamic processor models available in the Comp slot can also be loaded into the Gate slot.

The main parameters for the selected model are shown on the bottom of the screen and are adjustable by each corresponding knob below the main display.

The following parameters are available at the top of the screen:

- ACCENT: raises the gain temporarily when the gate opens to compensate for possible late gate opening.
- KEY SOURCE: select another channel as the sidechain input signal that will trigger the processor's response.
- KEY FILTER: apply a low-pass, high-pass, band-pass or notch filter to the Key Source signal.
- KEY SOLO: sends the Key Source signal to the Solo bus.
- KEY TAP: choose the point in the channel's signal path from which the Key Source signal is taken (see section 6.1 for details).

Other processors available in the Gate slot include a ducker, multiple analog gate emulations, compressors, Wave Designer, Auto Rider, a preamp emulation, Dynamic EQ, etc.

EQ

The EQ processing slot offers the standard WING EQ as well as analog emulations. The parameters shown on screen vary, depending on the EQ model chosen on the dropdown menu.

- FREEZE temporarily makes the RTA still until the button is clicked on again.

- MIX adjusts the amount of gain applied by each filter in the EQ. When at 100%, the amount of gain is applied as indicated on the EQ. Lower and higher percentages modify the applied gain proportionally.
- The LO CUT and HI CUT buttons enable/disable the filter in the INPUT slot.
- FILTER opens the filter section in the previous processing slot. When the WING EQ is selected, the low shelving and high shelving filters can be changed for parametric filters in the SETTINGS screen on the top right corner (Fig. 20). This screen also allows changing the EQ RTA settings.
- BAND SOLO sends the frequency range affected by the currently selected filter to the Solo bus.

COMP

The Comp slot offers the same dynamic processor models available in the Gate slot.

The main parameters for the selected model are shown on the bottom of the screen and are adjustable by each corresponding knob below the main display.

Besides the parameters provided in the EQ slot, the following are also available:

- GAIN: make-up gain after the compression
- XOVER MODE: the signal is divided into two paths. One path is affected by the selected filter (XOVER MODE) and compressed. The other path with the frequencies that were filtered out is then added back to the compressed signal. For example, when XOVER MODE is HI, the lower frequencies are filtered out of the signal to be compressed. The compressed higher frequencies are then added to the uncompressed lower frequencies.
- Note that the dB value indicated in the XOVER MODE refers to the slope around the filter's cutoff frequency. The maximum attenuation applied by the filter is set with the DEPTH parameter. FREQ defines the cutoff frequency.
- XOVER SOLO sends the post-filter signal to the Solo bus.
- AUTO ENV: when enabled, the ATTACK, HOLD and RELEASE parameters are adjusted automatically.
- DETECTOR: defines if the dynamic processor reacts to RMS or PEAK level measurements.

INS 1 and INS 2

The FX PROCESSOR dropdown menu allows you to assign an effect loaded on any of the 16 FX slots to the selected insert point. Empty slots can also be assigned. Once a slot has been selected, the effect loaded in that slot is chosen in the FX TYPE dropdown menu.

RESET restores all effect parameters to their default values.

FX SPILL temporarily assigns the effect parameters to the left fader section on WING and WING COMPACT, and to the 4 knobs on the front panel on WING RACK.

After loading an effect on the INS 1 or INS 2 slot of a Bus, Main or Matrix, "SET CUST" renames the channel with the effect name and assigns the "FX" icon to its scribble strip.

AUTOMIX

The INS 2 slot in the 40 INPUT channels offers the Automix option which can be enabled by toggling from FX to AUTO-X or AUTO-Y at the top of the screen (Fig. 64). This function is useful in situations such as panels with multiple speakers on stage since it automatically adjusts each channel's level to keep a relatively uniform output level and low noise floor when combining all the channels.

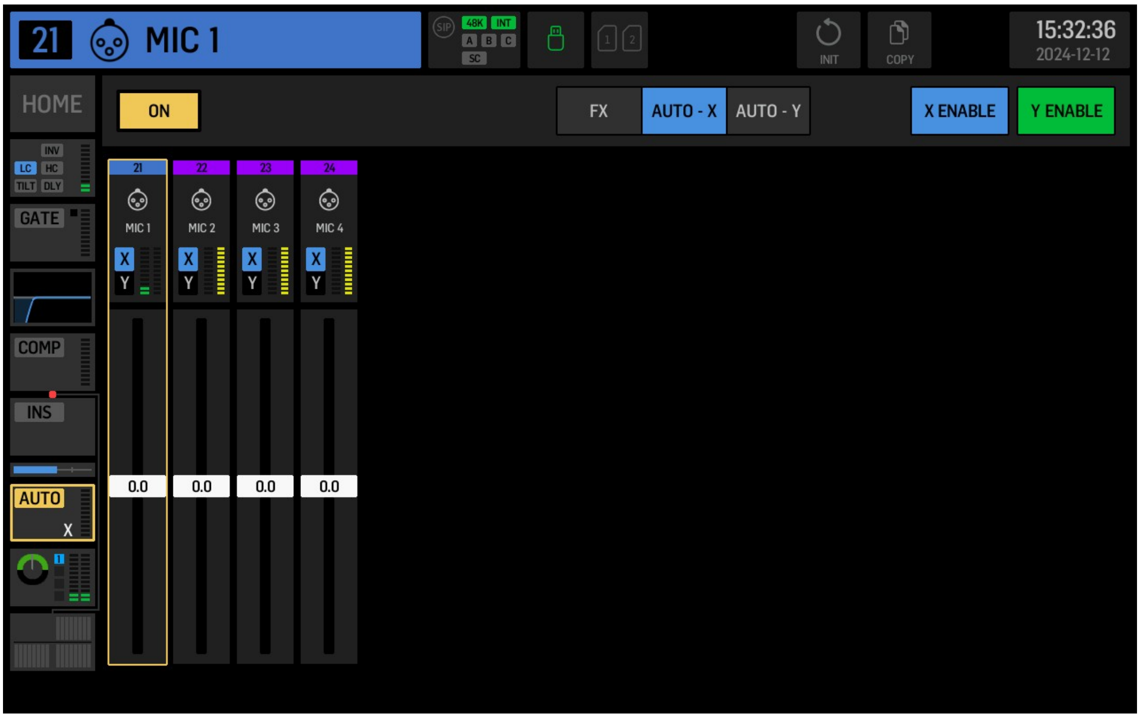

Figure 64: Automix

When Automix is enabled, an additional automatic attenuator is introduced after the channel fader. The loudest channel is either not attenuated or only slightly attenuated, while the other channels are attenuated more drastically to keep sound leakage and noise levels low. When two or more channels are active, they are attenuated proportionally so that the overall level does not exceed the level of a single active channel. Inactive channels remain drastically attenuated.

Since the last active channel is always kept on, there are no artifacts such as those created when noise gates open and close.

There are two independent Automix groups to which Input channels can be assigned: X and Y. Manual and fixed level adjustments can be made using the sliders on the screen. Note that adjusting the channel fader affects how Automix responds to that channel.

MAIN SENDS

The send level to each of the 4 Main buses is shown on the left side of the screen (Fig. 65). Each send can be set to the pre-fader position with the PRE button. The Pan and Width controls are shown on the center of the screen. The Width can be adjusted from -150% to 150%. Negative values invert the left and right channels.

A vector scope shows the phase correlation between the left and right channels (pre-panning) of the Source assigned to the channel. The channel fader, mute and solo buttons are shown to the right.

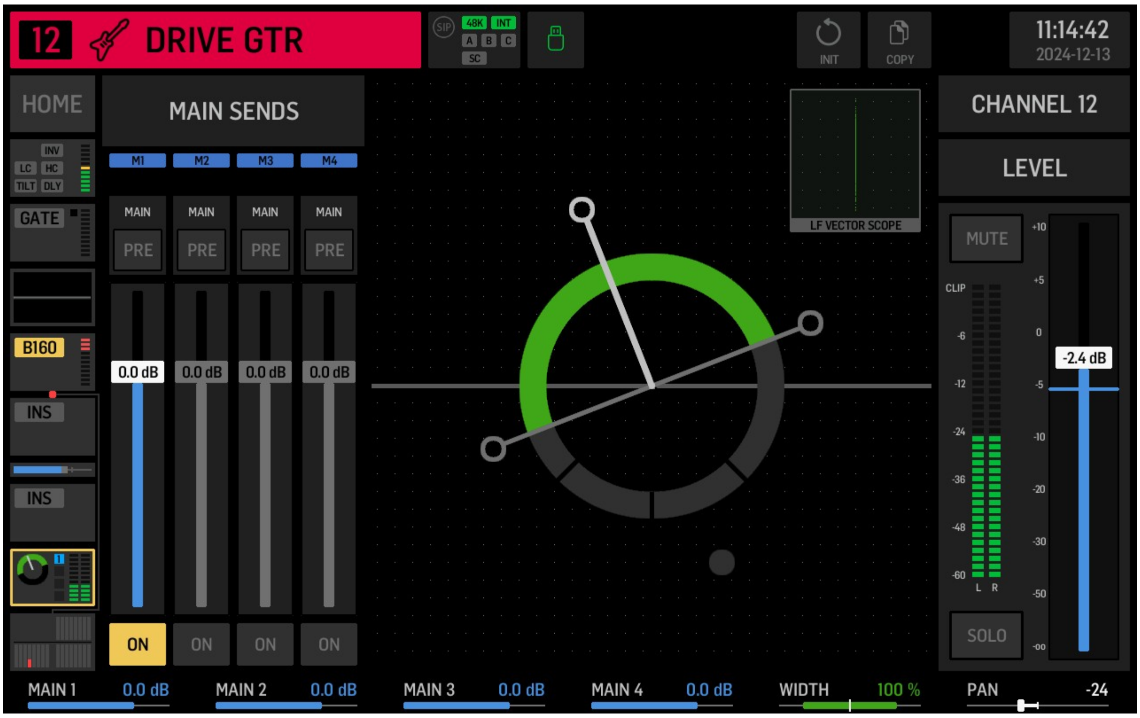

Figure 65: Main Sends

BUS/MATRIX SENDS

The Sends slot can open the Bus Sends (Fig. 66) or Matrix Sends (Fig. 67). The bottom half opens the Bus Sends and the top half opens the Matrix Sends. Alternatively, toggle between both screens by turning the knob for the PAGE parameter below the main display or by clicking on the Sends slot twice.

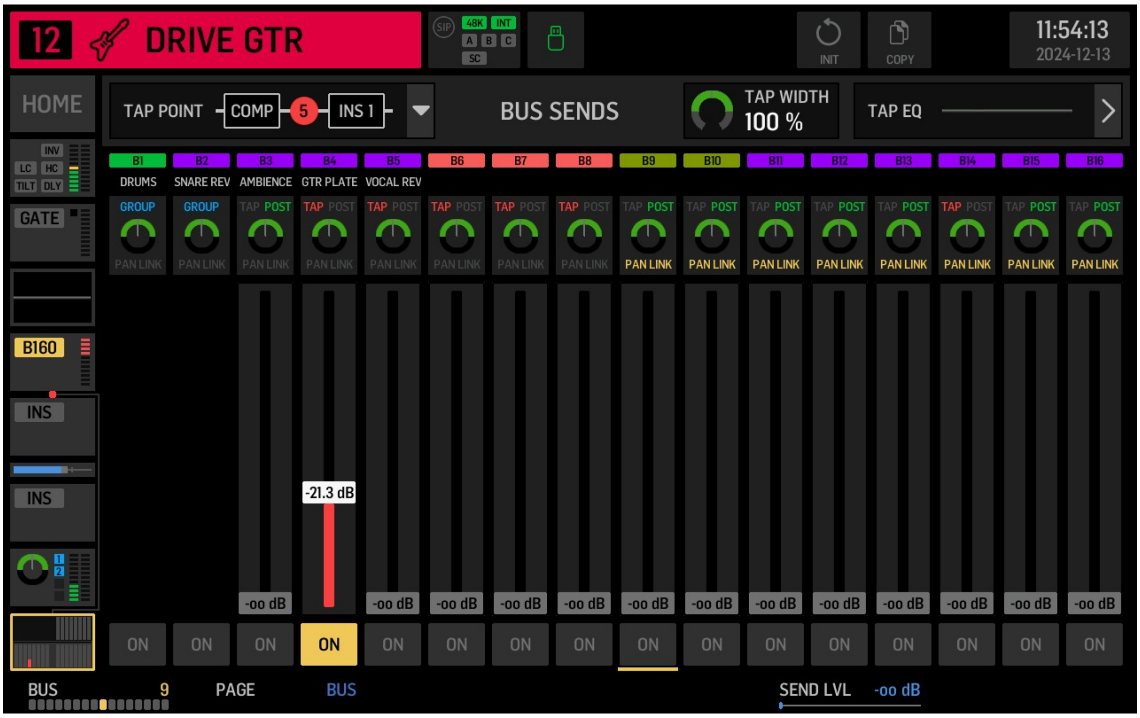

Figure 66: Bus Sends

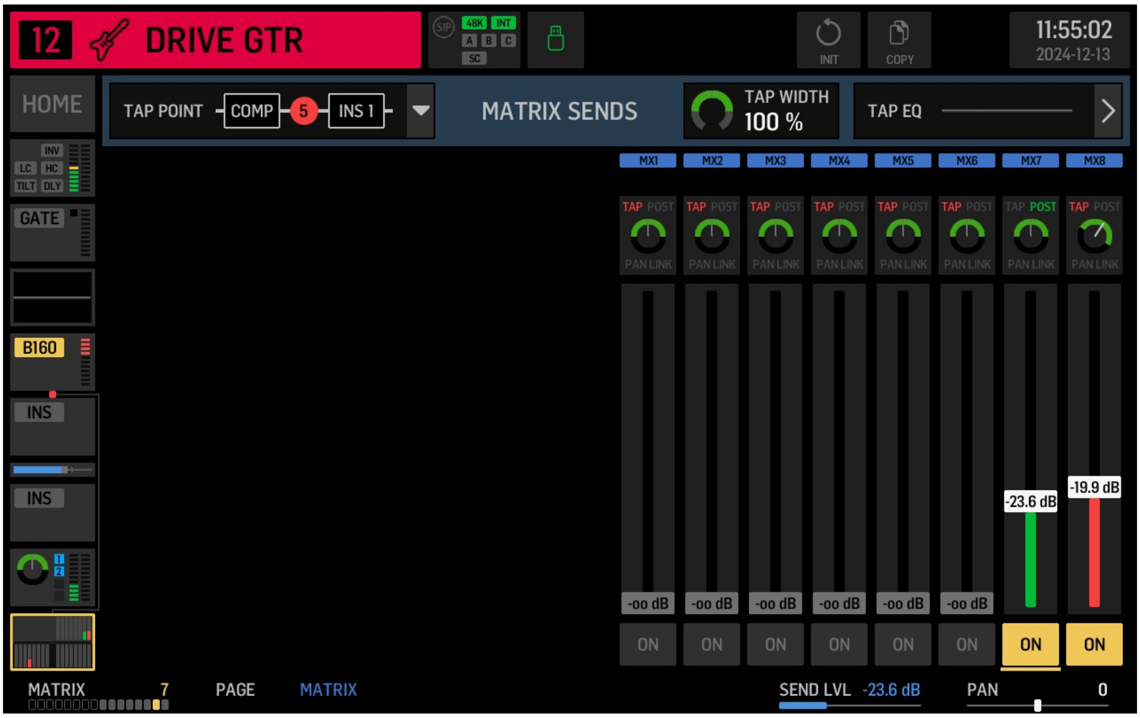

Figure 67: Matrix Sends

The upper left dropdown menu offers an alternate method for choosing the TAP point position, other than dragging and dropping the blocks in the Channel Home screen.

Click on the Pan knob to open a pop-up window with the following settings:

- MODE: sets the send mode from the selected channel to the corresponding bus. Note that the same BUS/MATRIX can receive signals from different channels set to different modes.
  - TAP: the signal is derived from the TAP point
  - POST: the signal is derived after the channel fader
  - GROUP: there is no separate send level control. The signal is sent after the channel fader.
- IGNORE CHANNEL MUTE (only available in TAP mode): the send remains active when the channel is muted.
- LINK SEND PAN TO CHANNEL PAN: the send panning is linked to the channel panning. When off, the send panning can be adjusted with the slider below.

TAP WIDTH determines the stereo width of the selected channel's sends currently set to TAP mode.

TAP EQ opens a separate screen with a 3-band equalizer applied to the signal sent in TAP mode.

6.2 Aux Channels

There are 8 Aux Channels available (Fig. 68). Each channel offers the following processing slots:

- INPUT slot

- MAIN/ALT input
- TRIM & BALANCE
- Pre-fader delay
- DYNAMICS slot (selected dynamic processors)
- 6-band EQ (same processors as on Input Channels)
- Pre-fader insert point
- MAIN SENDS
- BUS/MATRIX SENDS

The TAP point on Aux Channels is fixed post insert point and pre-fader.

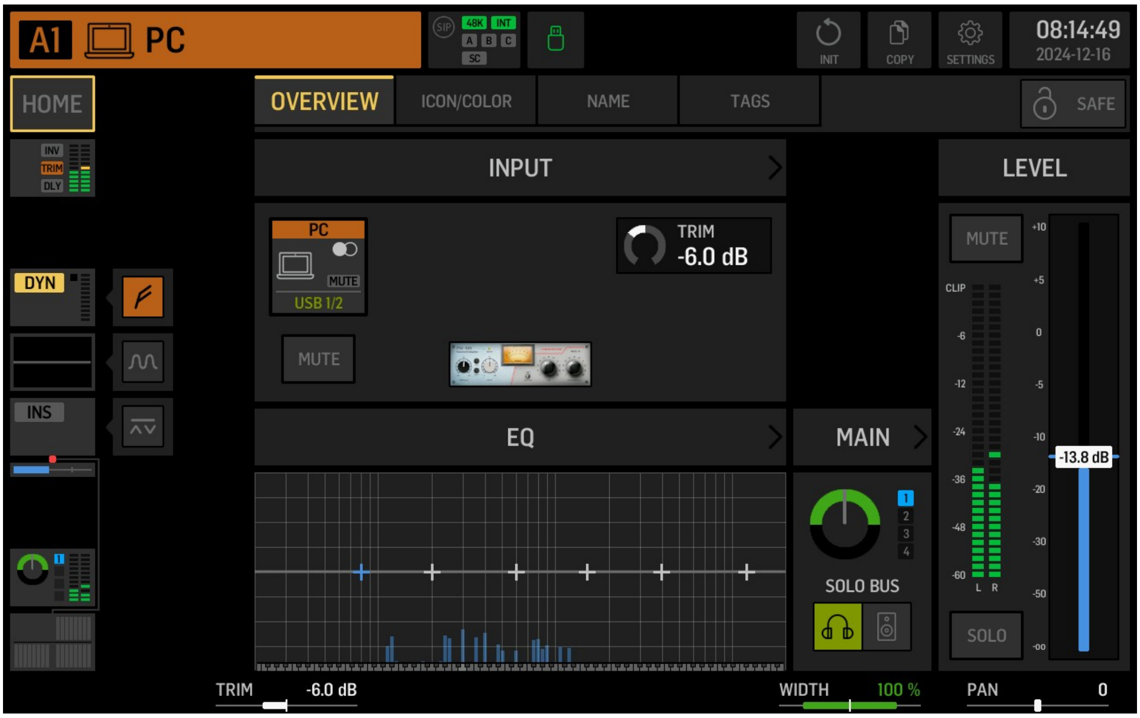

Figure 68: Aux Channel

6.3 Bus Channels

There are 16 Bus Channels available (Fig. 69). Each channel offers the following processing slots:

- Input slot
  - Bus Feed Configuration
  - Trim & Balance
- Post-fader delay
- Dynamics slot (same processors as Input Channels)

- Pre-fader insert point
- 8-band EQ (post-fader) (same processors as Input Channels)
- Post-fader insert point
- Main Sends
- Bus/Matrix Sends

The TAP point in Bus channels is fixed post insert point 1 and pre-fader.

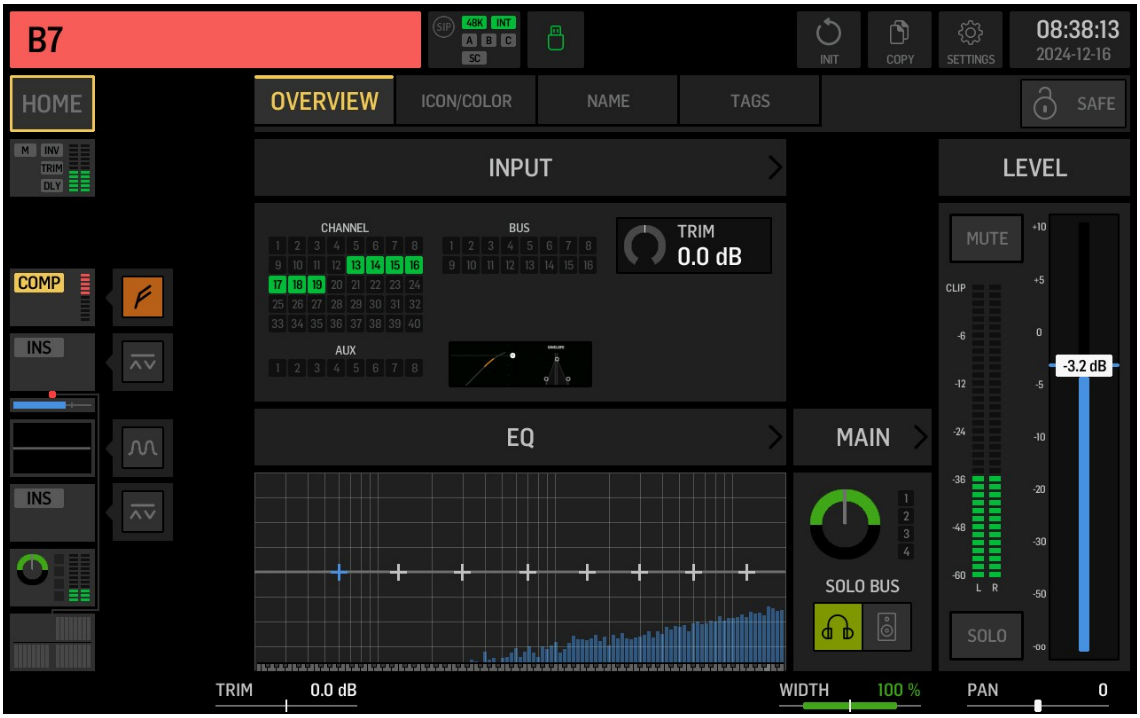

Figure 69: Bus Channel

6.4 Matrix Channels

There are 8 Matrix Channels available (Fig. 70). Each channel offers the following processing slots:

- Input slot
  - Matrix Feed Configuration
  - Trim & Balance
  - Direct Input: an additional signal (AES/EBU, Phones, Speakers or Solo bus) can be added to the selected Matrix bus
- Dynamics slot (same processors as Input Channels)
- Pre-fader insert point
- 8-band EQ (post-fader) (same processors as Input Channels)
- Post-fader insert point

- Output slot
  - Post-fader delay
  - Channel fader

Note that a Matrix can't be sent to other buses.

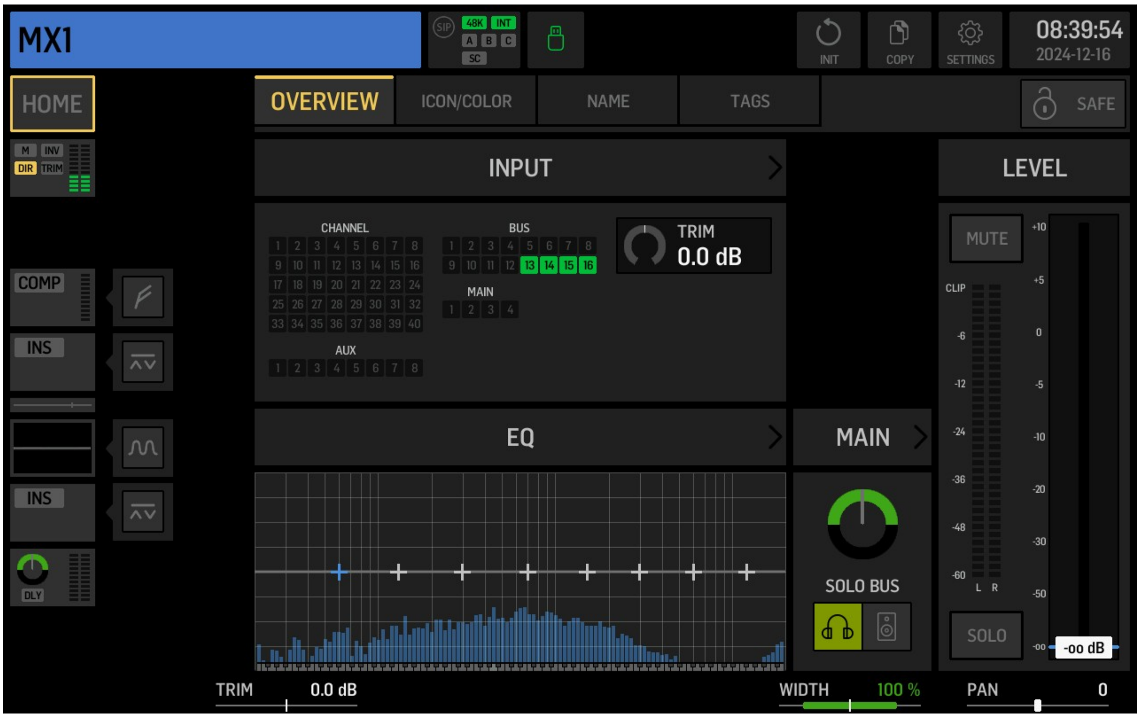

Figure 70: Matrix Channel

6.5 Main Channels

There are 4 Main Channels available (Fig. 71). Each channel offers the following processing slots:

- Input slot
  - Main Feed Configuration
  - Trim & Balance
- Dynamics slot (same processors as Input Channels)
- Pre-fader insert point
- 8-band EQ (post-fader) (same processors as Input Channels)
- Post-fader insert point
- Output slot
  - Post-fader delay
  - Channel fader
- Matrix Sends

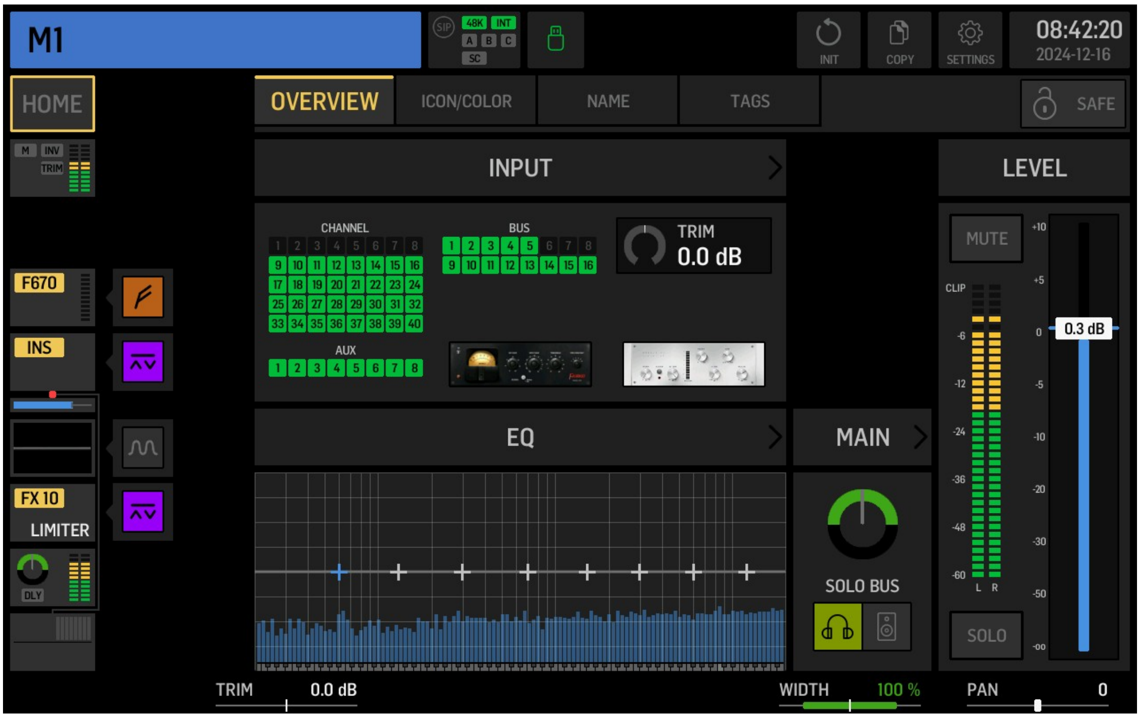

Figure 71: Main Channel

The sends from the Main Channels to any Matrix can be either pre- or post-fader.

6.6 Latency

When signals follow the conventional processing order, all channels remain aligned, and no latency compensation is required—regardless of any processing applied to the channels:

Input Channel → Bus → Main Bus → Matrix

Latency is introduced only when a signal is routed back to a previous stage in the chain. The following configurations introduce varying amounts of latency:

- Bus to Bus send: 8 samples
- Bus/Main/Matrix assigned as a Source on an Input Channel: 38 samples
- Insert Point: 32 samples

Certain effects introduce additional latency. The total latency for each effect is the sum of the insert point latency (32 samples) plus the following values:

Effect	Additional latency (samples)	
	44.1 kHz	48 kHz
STEREO PITCH (Delay set to 0*)	178	178
PRECISION LIMITER	48	52
EXCITER	10	11
PITCH FIX*	39	39
UK ROCK AMP	18	18
ANGEL AMP	18	18
JAZZ CLEAN AMP	14	14
DELUXE AMP	16	16
MASTERING	52	52

Table 6.2: Additional latency for effects in insert points

**STEREO PITCH minimum latency displayed. Actual latency may vary.*

Section 7

Monitors

The MONITORS screen (Fig. 72) provides multiple settings for controlling the headphones and speakers monitoring mixes, as well as the two talkback paths and the solo bus. To access this screen, press the VIEW button on the MONITORING/TALKBACK section or go to SETUP → MONITORS.

Figure 72: MONITORS screen

The left column contains the following controls:

- TALK A and TALK B buttons activate the corresponding talkback paths and are always in sync with the hardware buttons on the top panel (WING and WING COMPACT).
- PHONES/SPEAKERS/PH+SPK indicate the monitoring path that is currently active. Click on them to activate a different monitoring path.
- DIM attenuates the selected monitoring path by the amount set with the DIM knob.
- MUTE silences the selected monitoring path. Alternatively, hold down the hardware DIM button to enable MUTE.
- MONO sums the selected monitoring path to a mono signal.

- LR SWAP swaps the left and right channels. Alternatively, hold down the hardware MONO to enable LR SWAP.

The TALK A and TALK B tabs offer independent settings for each talkback path:

- SWITCH MODE determines how the talkback is activated with the TALK A and TALK B buttons
  - AUTO activates PUSH mode when the TALK A or TALK B button is held down and LATCH mode when the buttons are pressed and then released immediately.
  - PUSH activates talkback while the button is held down.
  - LATCH activates talkback until the button is pressed again.
- AUTO DIM determines the amount by which buses 1-16 (BUS DIM) and the selected monitoring path (MON DIM) are attenuated when talkback is active

The TALK TO BUS, TALK TO MATRIX and TALK TO MAIN grids allow you to select the buses to which the talkback signal is sent. TALK A and TALK B are independent.

When USE SEND LEVELS is active, the talkback signal is sent to the selected buses at the level set by the SEND levels from the talkback channel (CH40 or AUX8) to each bus. If disabled, the talkback signal is sent to all selected buses at a uniform level set by TALK LEVEL, irrespective of individual channel send levels.

The right section of the screen offers additional settings for both the PHONES and SPEAKERS monitoring paths. The set of tools are the same for both monitoring paths but independent of each other.

- SOURCE: choose the signal to be heard on the selected monitoring path. The following options are available: Main 1-4, Matrix 1-8, Bus 1-16, Aux 1-8
- SRC LEVEL: additional level control for the selected Source
- SRC MIX: when SOLO is active, the selected Source can still be heard at the level set by this knob.
- DIM: determines the amount by which the monitoring mix is attenuated when the DIM button is active.
- PFL DIM: determines the attenuation applied to the pre-fader solos.
- BAND SOLO: determines the boost applied when using the BAND SOLO function in the channel equalizers.
- DIRECT IN: selects an additional signal (Input or Aux channel, Bus, Main or Matrix) that is mixed into each monitoring bus.
- EQ: 8-band EQ for speaker/headphone correction, Pan and Width controls (Fig. 73)
- INVERT/DELAY
- LIMITER (threshold control)

Note that the level fader on the right side of the MONITORS screen is only controllable with the dedicated PHONES LEVEL and MONITOR LEVEL controls (on WING RACK, a CC can be used since there is no dedicated MONITOR LEVEL knob).

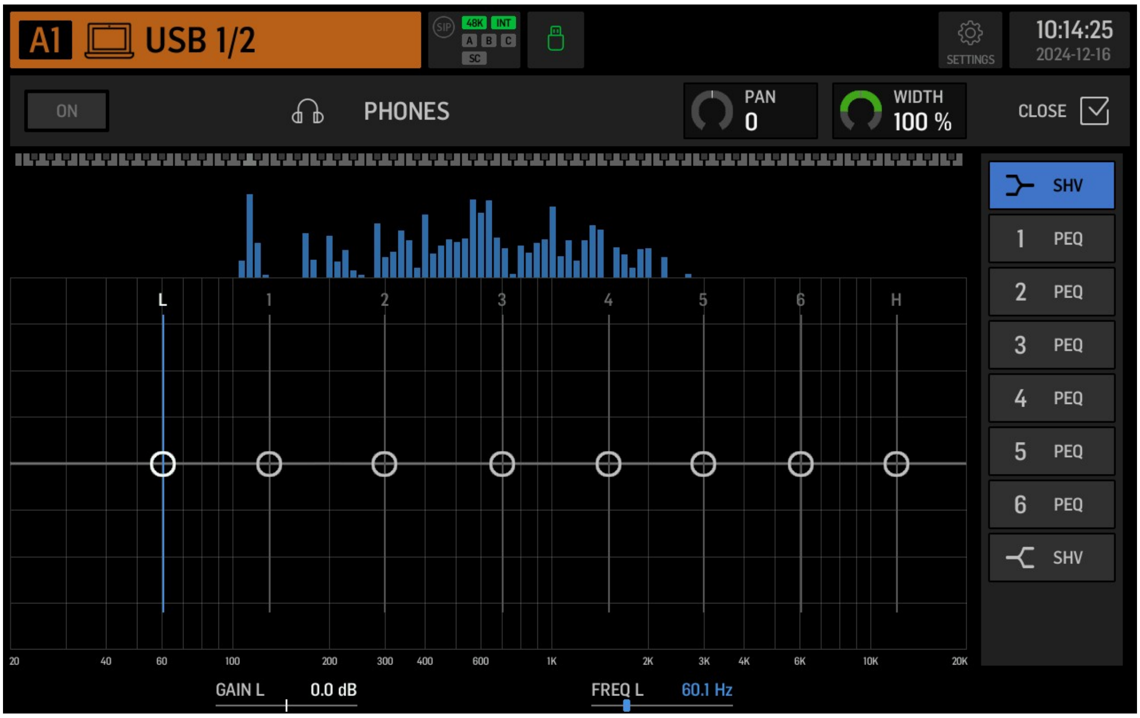

Figure 73: MONITORS EQ screen

Section 8

WING-LIVE and 4-Track USB Recorder

The WING-LIVE expansion card offers the possibility of recording and playing back 32 tracks to/from an SD card or 64 tracks to/from two linked SD cards. Similarly, 4 tracks can be recorded and played back to/from a USB stick connected to the front panel port.

Routing for recording

Go to ROUTING → OUTPUTS and select the output group "WLIVE REC" to route up to 64 channels to the WING-LIVE card; and "RECORDER" to route up to 4 tracks to the USB recorder.

Routing for playback

Go to ROUTING → SOURCES and select the SOURCE group "WLIVE PLAY" to route up to 64 channels from the WING-LIVE card playback; and "USB PLAYER" to route up to 4 tracks from the USB player.

WAV files (44.1/48kHz, 16/24 bit) can also be copied to the console's internal storage using WING DATA mode (see Section 5.5 → GENERAL) or to a USB stick. Note that WAV files in the internal storage can only be played back on the LIBRARY screen (see Section 5.6 → CLIP).

8.1 WING-LIVE

WING can record up to 64 channels onto two linked SD cards. Each card records multitrack WAV audio files of 8, 16 or 32 channels called a session. A session can be separated into individual files for each channel using the LiveSessions app available on behringer.com. LiveSessions can also convert Individual WAV files to session to be played back on the console.

Note that the SD cards must be formatted to FAT32 with an allocation unit size (cluster) of 32 kB. SD cards of up to 32 GB can be formatted directly on the console (see SETTINGS section below).

When recording, WING-LIVE automatically creates a folder named "X_LIVE" in the SD card root directory. Within the X_LIVE folder, individual folders are created for each session. When creating a session with the LiveSessions app, an X_LIVE folder is also created. Make sure to copy it to the SD card root directory.

The WING-LIVE main screen (Fig. 74) provides transport controls for the SD recorder and metering information for the 32 channels routed to each SD card.

Clicking on the folder icon shows the sessions on the left. Clicking on the cursor icon shows the markers on the open session.

It is possible to adjust the cursor position on the timeline at the bottom of the screen by dragging it or using the knob below it.

When recording more than 32 channels onto SD cards, both SD cards must be linked with the

LINK CARDS button.

Figure 74: WING-LIVE Recorder

Clicking on the right arrow button to the right of the SD CARD 1 or SD CARD 2 header opens the card settings screen (Fig. 75).

- SD RECORD: select the number of channels to be recorded (8, 16 or 32)
- MODE:
  - PLAY (yellow highlight): the session is played back normally
  - A→B (blue highlight): the user can define the start and stop points with the knobs below the main display. Playback stops when reaching the stop point.
- LOOP (green highlight): the session is played back in loop. The loop borders are defined with the knobs below the main display.

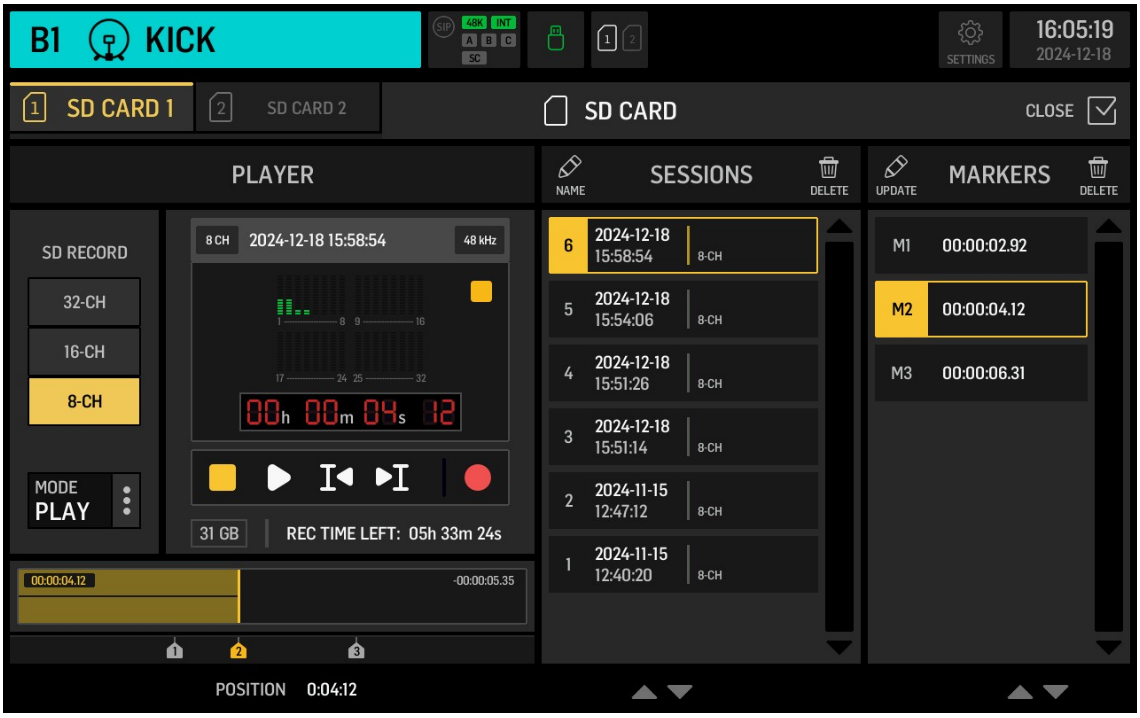

Figure 75: WING-LIVE SD card settings

When a session is being played back, markers can be added on this screen with the following button:

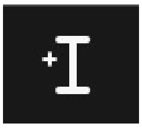

Figure 76: Add marker

Use the following buttons to navigate to the previous and next markers:

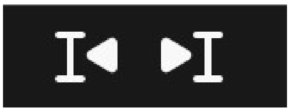

Figure 77: Marker navigation buttons

The existing sessions and markers are displayed in their respective columns. Sessions can be renamed, and the markers' positions can be changed by adjusting the cursor position, selecting the desired marker and clicking on UPDATE.

Settings

When on the WING-LIVE recorder screen, a SETTINGS button is displayed on the top right corner of the screen. The following settings are available:

- SHOW METERS: shows or hides the level meters of the incoming and outgoing signals to/from the SD multitrack recorder.
- AUTO INPUT SELECT: determines which of the two SD card sections controls the automatic

MAIN/ALT input selection on the channels routed to and from the WING-LIVE recorder (WLIVE REC and WLIVE PLAY output and input groups). The automatic behavior is defined by the following three parameters and only applied to channels selected in the Global Input Select Enable screen (Fig. 35)

- STOP / PLAY / RECORD INPUT: keeps the channels' SOURCE or switches it to MAIN or ALT depending on the WING-LIVE stop, play or record status.
- FORMAT: formats the selected SD card

Battery Backup

A 3V CR123 battery can be installed in the WING-LIVE card to allow the multitrack recording to be stopped and saved in case of a power failure.

To install the battery:

1. Turn off and unplug the console
2. Remove the WING-LIVE card from the console
3. Insert the battery into the black plastic holder. Make sure to match the polarity printed on the plastic holder.
4. Install the WING-LIVE card back in the console.

8.2 4-Track USB Recorder

Access the USB Recorder (Fig. 78) by clicking on the USB stick icon at the top of the screen.

Create an empty playlist (Fig. 79) by clicking on the CREATE PLAYLIST button. Alternatively, create a playlist with the selected clip by clicking on the ADD button. If a folder is selected, it is possible to add all the clips in that folder to the playlist with the ADD ALL button.

Once a playlist has been created, the icon shows how many clips (SONGS) are in it. The following options are available on the PLAYLIST screen:

- REPEAT: the playlist is on loop. After the last clip is played, the first one is played again.
- REMOVE: remove the selected clip from the playlist.
- SKIP: temporarily ignores the selected clip from the playlist until SKIP is disabled again.
- STOP: stop the playlist after the selected clip ends. Hold down the button to apply to all clips in the playlist.
- UP/DOWN: move the selected clip.

Use the SAVE, CLOSE and OPEN PLAYLIST buttons for saving and recalling playlists.

The recorder options are located in the top-right panel. It supports recording 2- or 4-track WAV files at 16- or 24-bit resolution. The sample rate is determined in the SETUP → AUDIO screen.

The storage location of the recorded files on the USB drive can be specified by navigating to the desired location and clickin on REC PATH. The active recording location is displayed to the right, above the "4-TRACK RECORDER" text.

- REC: starts the recording

Figure 78: USB Recorder screen

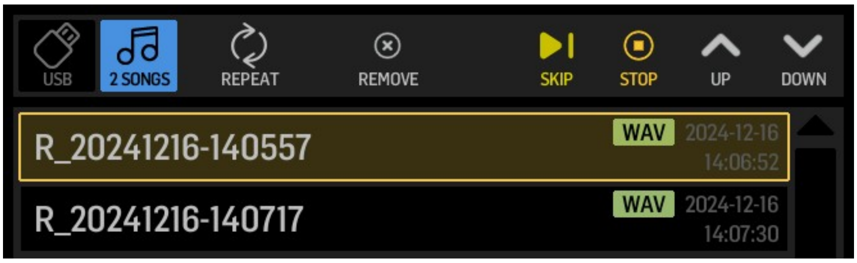

Figure 79: Playlist

- PAUSE: pauses the recording without creating a new file
- STOP: stops the recording. Starting REC again creates a new file.
- NEW FILE: creates a new file without interrupting the recording

Playback options are available on the bottom right panel. It is possible to drag the playback cursor when paused. Use the “←” and “→” buttons to navigate to the previous and next clips in a playlist.

Section 9

Firmware Update

The WING console firmware can be easily updated with a USB flash drive. Download the firmware file **specific to your WING model** (WING, WING COMPACT or WING RACK) from the product page on behringer.com and follow these steps:

1. Copy the firmware file to a FAT32-formatted USB flash drive and connect the drive to the USB port on the top panel.
2. Go to SETUP → GENERAL → UPDATE.
3. Select the desired firmware file and click on UPDATE.
4. Reboot the console by going to SETUP → SHUTDOWN → CONFIRM and then holding down EFFECTS and pressing HOME.

The firmware can also be updated using a computer connected via USB:

1. Open the SETUP → GENERAL page and click on WING OS.
2. Connect the console to a computer with a USB cable.
3. A virtual drive named "WING OS" will appear. Open the drive.
4. Copy the new firmware file onto the drive. It is recommended to delete older firmware files from the WING OS drive or move them to a subfolder.
5. Eject the drive from your computer.
6. Reboot the console by going to SETUP → SHUTDOWN → CONFIRM and then holding down EFFECTS and pressing HOME.

Emergency Boot Mode

If the console does not boot up normally, for example if a wrong firmware file is loaded on the console, you can still access the WING OS and WING DATA drives to load the correct firmware file:

1. With the console powered off, connect it to your computer with a USB cable.
2. Press and hold the blank button to the right of the main display (SETUP on WING RACK), then power the console on.
3. Two drives named "WING OS" and "WING DATA" will appear on your computer.
4. Drag the new firmware file into the WING OS drive.

5. Eject the drive from your computer.

6. Reboot the console by going to SETUP → SHUTDOWN → CONFIRM and then holding down EFFECTS and pressing HOME.

Section 10

Shortcuts

Purpose	Activated by	Exit	Notes
Deactivate main display touch control	Hold SETUP, blank button (UTILITY on WING RACK) and CLR SOLO, until a small X is shown in the status bar of main display	Hold blank button (SETUP on WING RACK) and CLR SOLO until the X disappears	No touch events will have any effect, while the small X is shown. The console will keep operating as usual otherwise. The touch screen is calibrated again after exiting this mode and is operational again.
Reset touch panel	Hold blank button (SETUP on WING RACK) and CLR SOLO for longer than 1.5s		Resetting the touch panel can help to temporarily fix ghost touch issues
Ghost click test	Hold METERS and HOME for 5s while powering the console up	Turn off the console	
Emergency boot mode	Hold blank button (SETUP on WING RACK) while powering up the console	Eject the WING OS and WING DATA drives from the computer	In case the wrong or a faulty firmware file is loaded on the console and it does not boot, this mode gives access to the WING OS and WING DATA drives to transfer firmware files and scenes, respectively.
Restart console	SETUP → SHUTDOWN. Then hold EFFECTS and press HOME.		

Purpose	Activated by	Exit	Notes
Surface Test Mode	Hold blank button (SETUP on WING RACK) while powering the console up		Tests all LEDs on the control surface.
Surface Lock	Hold HOME. Alternatively, hold HOME together with any other combination of buttons to serve as “password”.	Hold HOME or HOME and the previously determined button combination. Alternatively, restart the console manually.	Locks the console’s control surface and touchscreen, without affecting the audio or remote-control functions. All the buttons to the left of the main display can be used for locking/unlocking the console.
Initialize console (only temporary storage, no snapshots or any other scenes are erased)	Hold CLR SOLO while powering the console up		Same as using SETUP → INITIALIZE CONSOLE but this mode acts before the console loads any saved state (in case the last loaded scene is leading to issues).
Take screenshot	Hold CLR SOLO then press UTILITY		Stores a bmp file of the current screen on a USB flash drive. A folder named “screens” must be created on the USB drive’s root folder beforehand.
Bypass automatic load of startup files	Hold LIBRARY while powering the console up		Does not load STARTUP*.snap, STARTUP*.snip and STARTUP*.show files in the STARTUP directory during boot
Configuring optional hardware, i.e. internal AoIP modules for Dante or WSG	Hold UTILITY for 5s while powering the console up		Enter “MOD-WSG” for the internal SoundGrid module or “MOD-DANTE” for the internal Dante module. Enter “MOD-NONE” if the module has been removed.
Adjust main display brightness	Hold blank button (CLR SOLO on RACK) and turn the knob under the main meter		

Section 11

Internal Effects

This section details all internal effects available on the WING console, including those found in the DYN, EQ, and COMP slots of individual channels, as well as those available exclusively through the FX Rack. Some effects are accessible in both the channel processing slots and the FX Rack.

11.1 Equalizers

All equalizers have a MIX control. At 100%, the EQ curve is applied exactly as shown. Lower values reduce the gain adjustments proportionally across all bands, while higher values intensify the gain changes beyond the displayed settings.

WING EQ

WING EQ is the console's default digital equalizer, featuring six parametric filters.

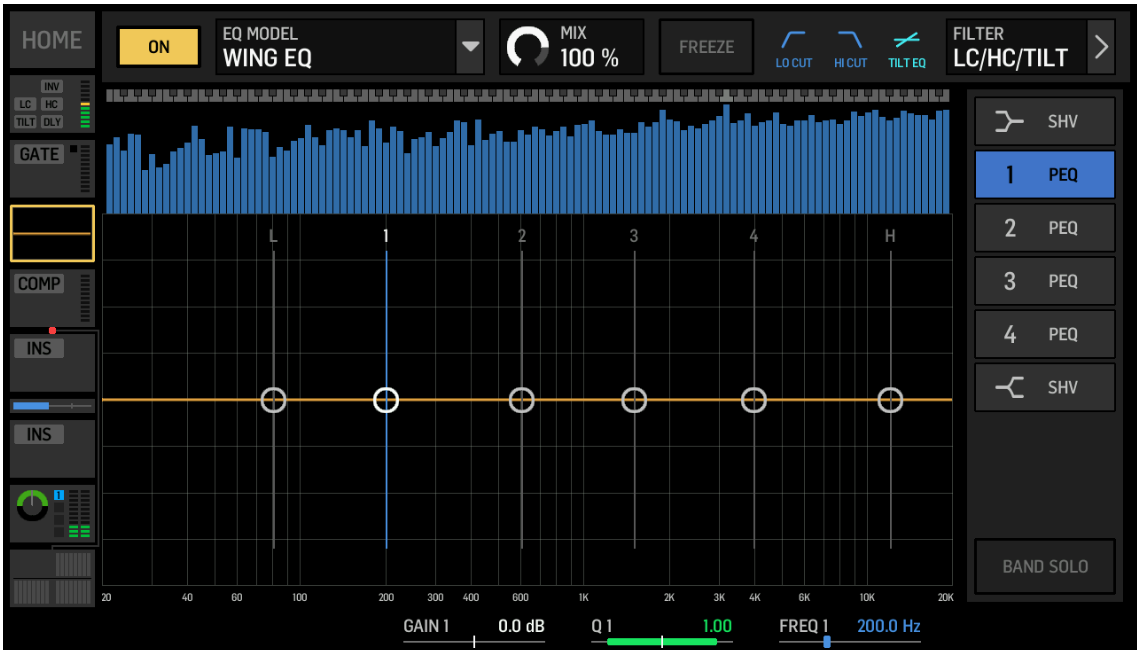

Figure 80: WING EQ

The two outer filters (L and H) are configured as shelving filters by default. To change them to parametric filters, tap SETTINGS to access the EQ/RTA settings screen (Fig. 25).

The EQ can be bypassed using the ON/OFF switch. When active, the EQ curve is displayed in yellow.

Each filter is adjusted individually by selecting it from the filter list on the right side of the screen. Once selected, its parameters become available for modification in the lower section of the display.

Hold down the button corresponding to a filter to reset it.

Each filter can be adjusted directly on the touchscreen. Drag vertically to change the gain, drag horizontally to change the frequency, and drag horizontally with a second finger to adjust the Q.

The LO CUT, HI CUT, and TOOL FILTER (TILT EQ, MAXER, AP 90, or AP 180) located in the INPUT slot operate independently from the EQ slot, but their effect is also represented in the EQ graph for reference.

Parameter	Function
GAIN	Amount of amplification or attenuation applied by an EQ band
Q	Width of the frequency range affected by a parametric filter
FREQ	Center frequency of a parametric filter or cutoff frequency of a shelving filter

Graphic EQ

A standard 31-band graphic equalizer covering the frequency range from 20 Hz to 20 kHz. Each band allows up to ±15 dB of gain adjustment.

Drag horizontally on the bottom part of the yellow selection to move it.

Figure 81: Graphic EQ

Enabling the TRUE CURVE function minimizes interaction between overlapping filters. This avoids the gain from adjacent filters from being combined also prevents “ripples” in the EQ curve.

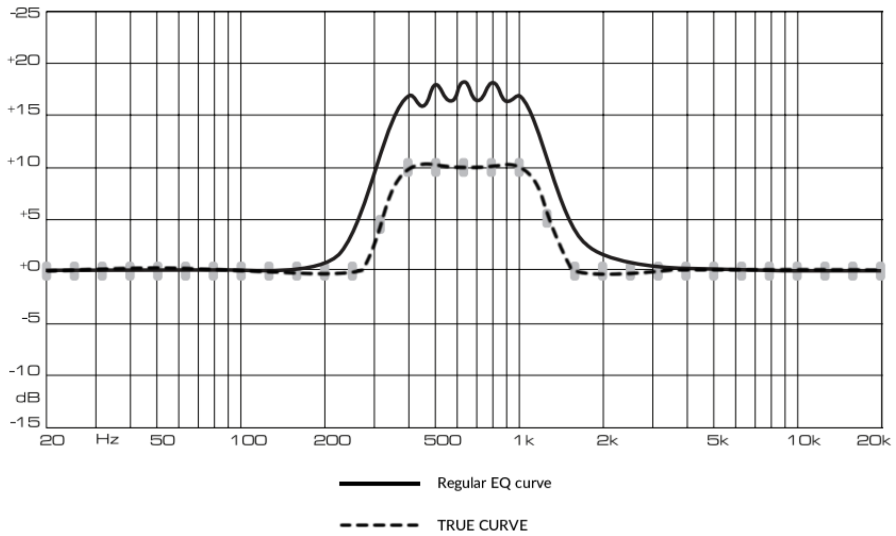

Figure 82: TRUE CURVE

PIA 560 GEQ

Modeled after a 500-series graphic EQ, this EQ features 10 bands spaced one octave apart. Each filter's bandwidth becomes narrower as the level of boost or cut is increased.

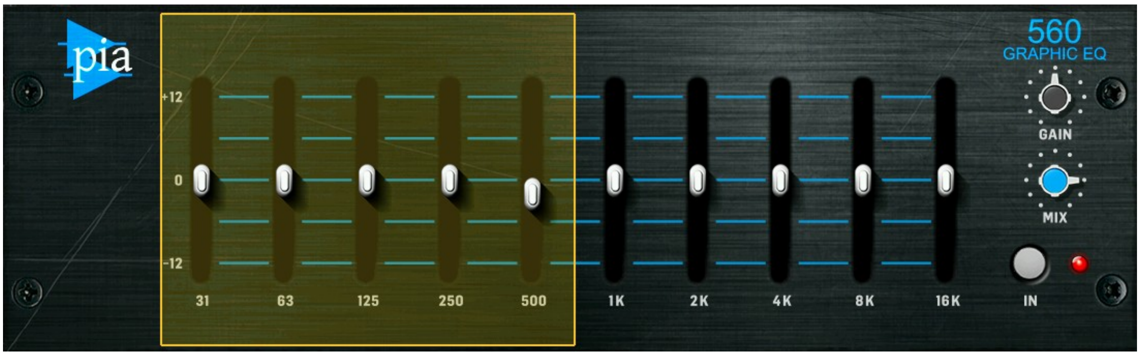

Figure 83: PIA 560 GEQ

Triple DEQ

Three-band dynamic equalizer with individually adjustable bands, accessed via the DEQ 1, 2, and 3 buttons. Each band offers a dedicated set of parameters. A single-band version of this processor is also available in the GATE slot of each channel.

Triple DEQ functions as a flexible multiband dynamics tool, depending on the Gain setting and the Above/Below switch:

- Negative Gain + Above: Multiband downward compressor
- Negative Gain + Below: Multiband downward expander
- Positive Gain + Below: Multiband upward compressor
- Positive Gain + Above: Multiband upward expander

Figure 84: Triple DEQ

Parameter	Function
Attack	How quickly the processor applies the gain adjustment after the signal crosses past the threshold—either above or below, depending on the setting of the Above/Below switch
Release	How quickly the processor stops applying gain adjustment after the signal crosses back past the threshold—either below or above, depending on the setting of the Above/Below switch
Threshold	Level the signals must exceed for the equalizer to engage
Gain	Maximum gain adjustment applied by each band
Freq	Center frequency for parametric filters or cutoff frequency for shelf filters
Q	Width of frequency range affected by parametric filters only. This parameter has no effect on other filter types
Ratio	Reduces output level relative to how much the input exceeds the threshold. For example, at a 2:1 ratio, every 2 dB past the threshold results in only a ±1 dB change at output. The maximum gain adjustment is determined by the Gain parameter.
Filter Types	Bands can be configured as a low or high Shelf filter, Parametric, or Flat. When Flat is selected, the band functions as a full-range dynamics processor. For shelving filters, 6 dB or 12 dB per octave selectable slope options of are available.
Above / Below	Determines whether the unit applies the gain adjustment when the input signal exceeds the threshold (Above) or falls below the threshold (Below)

SOUL Analog

Four-band parametric EQ based on a British console, known for its EQ module's subtle enhancement of both low and high frequencies. This effect is done by narrowing the Q factor of the low-mid and high-mid bands when gain is increased. Both center frequencies can be modified using the ÷3 and ×3 buttons.

Figure 85: SOUL Analog

Parameter	Function
EQ IN	Toggles processing on or off
÷3	Divides the low-mid filter center frequency in kHz by 3
×3	Multiplies the high-mid filter center frequency in kHz by 3

Even 88-Formant

Four-band parametric EQ modeled after the equalizer section of a British console series introduced in the early 2000s. This EQ is known for its rich low-end and airy high frequencies. The LOW band (black) and HIGH band (green) can be toggled between parametric and shelving modes. The LOW MID band is adjusted using light-blue controls, and HIGH MID band by adjusting using blue controls.

Figure 86: Even 88-Formant

Parameter	Function
EQ	Toggles processing on or off

Parameter	Function
Low Shelf On	Switches the LOW band between parametric and shelving filter types
High Shelf On	Switches the HIGH band between parametric and shelving filter types
HI-Q	Toggles the Q factor of the LOW and HIGH bands between 0.7 when inactive and 2.0 when active

Even 84

Three-band EQ modeled after the equalizer section of a British preamp and EQ, which shared the classic 1073 design and also provided selectable frequencies. The two shelving filters are part of a shared sixth-order filter circuit. Meaning, adjusting one filter can influence the frequency response of the other.

Like the original hardware, EVEN 84 includes a HI-Q button that narrows the MID bandwidth.

The three bands consist of two concentric knobs each. The inner knob controls the gain, while the outer knob controls the frequency.

Figure 87: Even 84

Parameter	Function
GAIN	Adjusts output level
LOW	Sets low-shelf filter frequency (off, 35 Hz, 60 Hz, 110 Hz, 220 Hz) and gain (±15 dB)
MID	Sets semi-parametric filter frequency (off, 350 Hz, 700 Hz, 1.6 kHz, 3.2 kHz, 4.8 kHz, 7.2 kHz) and gain (±20 dB)
HIGH	Sets high-shelf filter frequency (off, 10 kHz, 12 kHz, 16 kHz) and gain (±20 dB)
HI-Q	Increases the Q factor of the MID filter, making it narrower
EQ	Toggles the processing on or off

Fortissimo 110

Four-band parametric equalizer modeled after the EQ section of a British console, originally designed in 1985. The low and high shelving filters use higher-order circuits, enabling distinctive tonal shaping characteristics typically associated with enhancers.

The two parametric bands offer controls for gain, frequency, and Q. The low and high shelving bands provide gain and frequency controls.

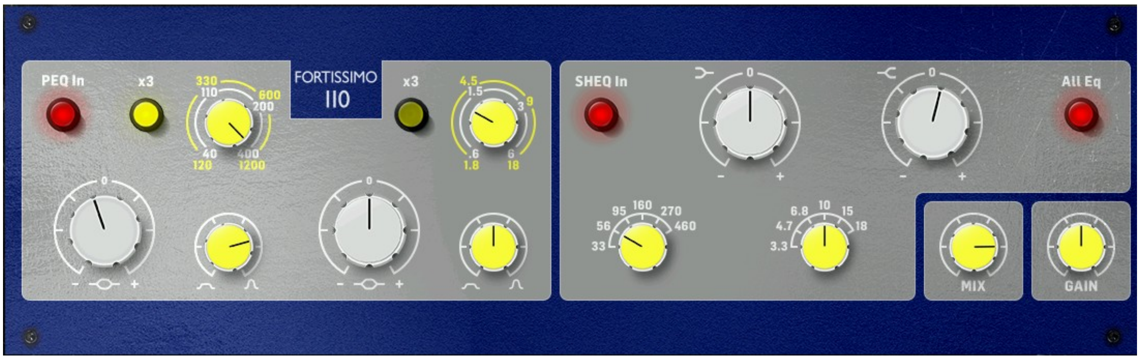

Figure 88: Fortissimo 110

Parameter	Function
PEQ In	Toggles two parametric filters on or off
SHEQ In	Toggles two shelf filters on or off
×3	Multiplies the low-mid and high-mid filters' center frequency by 3
All Eq	Toggles processing on or off

Pulsar

The PULSAR P1a/M5 is inspired by the passive tube equalizers introduced in the 1950s, which are renowned for their wide and gentle curves resulting in round and gentle tone-shaping.

These are passive equalizers, meaning their filters only attenuate frequencies. A tube-based amplifier is used to restore the overall signal level. To "boost" a specific band, the unit attenuates all surrounding frequencies while leaving the target band unaffected. It then amplifies the entire signal at the output stage. This method makes this EQ more suitable for overall tone adjustments, rather than precise changes.

As with the original hardware, the actual frequency ranges affected by the filters often extend beyond the values marked on the knobs and change depending on whether the band is boosted or cut. When both boosting and cutting the lower band simultaneously, a characteristic dip effect occurs just before the low-end boost.

The EQ consists of the following filters:

- MID EQ5
  - LO-MID BOOST: semi-parametric
  - MID DIP: semi-parametric
  - HI-MID BOOST: semi-parametric
- HI-LO EQ1
  - LO ATT: low-shelf

- LO BOOST: parametric
- HI BOOST: parametric
- HI ATT: high-shelf

Figure 89: Pulsar P1a/M5

Parameter	Function
MID EQ5 IN	Toggles semi-parametric low-mid, mid and high-mid filters on or off
LO-MID FREQ	Selects low-mid center frequency in Hz
LO-MID BOOST	Adjusts gain boost applied to the low-mid band
MID FREQ	Selects mid center frequency in Hz/kHz
MID DIP	Adjusts attenuation applied by the mid band
HI-MID FREQ	Selects high-mid center frequency in kHz
HI-MID BOOST	Adjusts gain boost applied by the high-mid band
HI-LO EQ1 IN	Toggles the low shelf, high shelf, and high parametric filters on or off
LO ATT	Adjusts attenuation applied by the low band
LO FREQ	Selects low band cutoff frequency in Hz
LO BOOST	Adjusts gain boost applied by the low band
HI WIDTH	Adjusts width of the high parametric filter. Lower values produce a narrower filter, while higher values result in a broader frequency range.
HI FREQ	Selects center frequency of the high parametric filter in kHz
HI BOOST	Adjusts gain boost applied by the high parametric filter
HI ATT	Adjusts attenuation applied by the high shelf filter
HI-ATT FREQ	Selects cutoff frequency of the high shelf filter

Mach EQ4

Six-band fixed-frequency equalizer with minimal phase shift, modeled after hardware known for its clean and detailed sound. The AIR FREQ control adds natural brightness, while the AUTO function compensates for gain changes introduced by the EQ automatically.

Figure 90: Mach EQ4

Dynamic EQ

A single-band version of the Triple DEQ (Fig. 84). It is available in the Gate and Comp slots of Input Channels.

In this effect, the Key Solo function is replaced by Band Solo, which isolates the selected band.

Dual Dynamic EQ

A dual-band version of the Triple DEQ (Fig. 84). It is available in the Gate and Comp slots of Input Channels.

In this effect, the Key Solo function is replaced by Band Solo, which isolates the selected band when the Gain parameter is set to a value other than 0 dB.

Figure 91: Dual Dynamic EQ

11.2 Dynamics

Common Controls

The following controls are available on multiple dynamic processors, enabling features such as parallel compression and filters in the sidechain signal:

Parameter	Function
KEY SOURCE	Select which sidechain signal the compressor will respond to
XOVER MODE	Apply a filter to the signal that is fed into the compressor. The frequencies removed by the filter are reintroduced after compression to retain their original dynamics
XOVER SOLO	Listen to filtered-out frequencies on the SOLO bus
KEY FILTER	Apply a high-pass, low-pass or band-pass filter to the sidechain signal
KEY SOLO	Listen to sidechain signal on the SOLO bus
DETECTOR	Selects whether sidechain signal is detected based on its PEAK level or RMS (average) level
ENVELOPE	Determines whether gain change is applied linearly (LIN) or logarithmically (LOG) over time; as specified by the Attack, Hold, and Release settings
AUTO ENV	Attack, Hold and Release settings are adjusted automatically
GAIN	Post-processing gain to compensate for the overall attenuation caused by the compression (Make-Up Gain)

WING Compressor

The default digital compressor on the console. A transfer curve graph on the left side of the display provides a visual representation for the input and output signals and when gain reduction is applied.

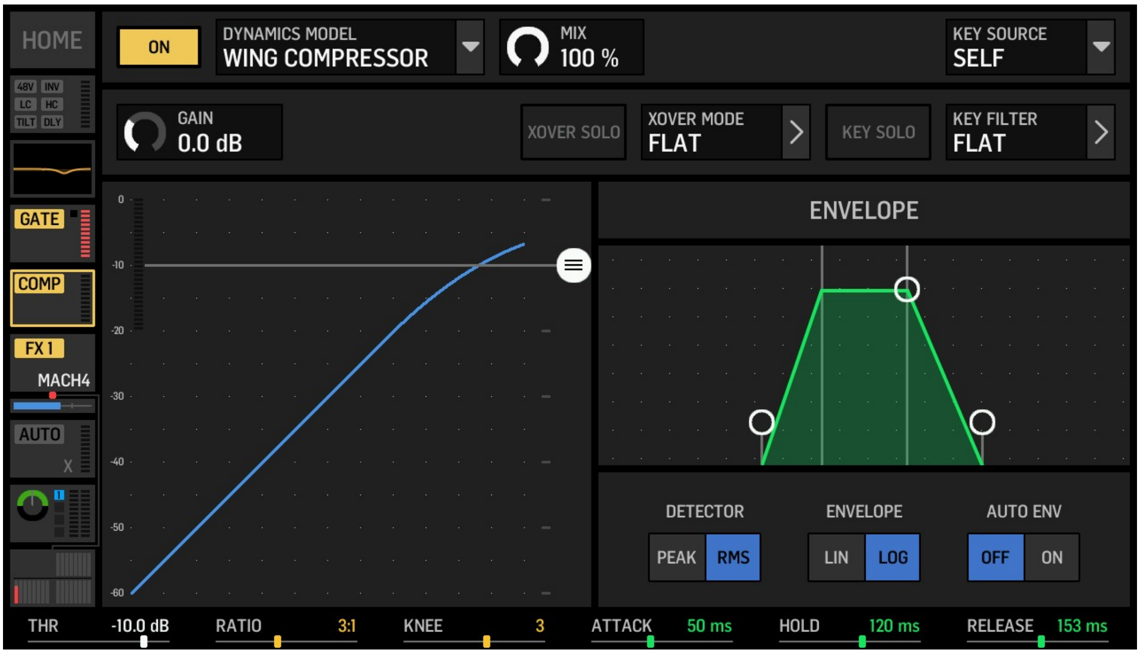

Figure 92: WING Compressor

Parameter	Function
THR	Threshold in dB, when reached, gain reduction is applied
RATIO	The relationship between the input signal's excess over the threshold and the output signal's excess over the threshold
KNEE	When active, Ratio increases around the Threshold, starting from 1:1:1 and gradually reaching the user-selected Ratio. Higher values result in a wider transition range and a smoother transition
ATTACK	Time it takes for the compressor to apply approximately two-thirds of the targeted gain reduction. Compressor becomes active when the input signal either: 1. Exceeds the threshold 2. Is already above the threshold and continues to rise, increasing the targeted amount of gain reduction
HOLD	Amount of time in ms until the release phase is engaged
RELEASE	Defines time taken to restore approximately two-thirds of the applied gain reduction. Becomes active when the input signal either: 1. Falls below the threshold, restoring gain toward unity 2. Decreases while still above the threshold, reducing the targeted amount of gain reduction

WING Expander

A conventional downward expander. Expander is available in the COMP processing slot.

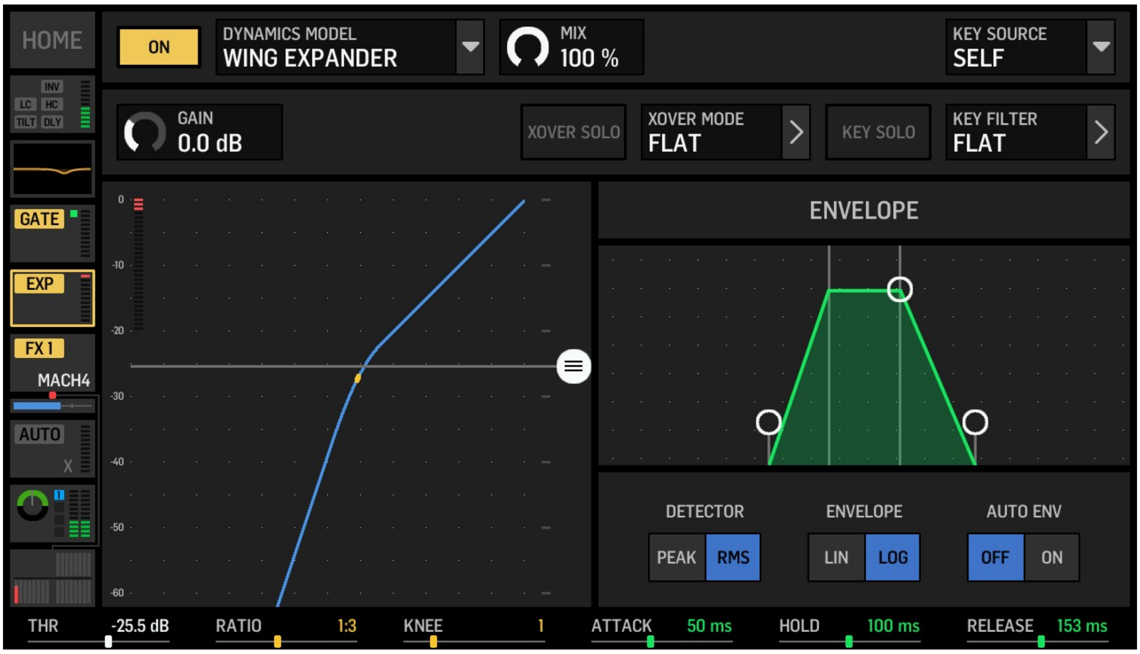

Figure 93: WING Expander

Parameter	Function
THR	Gain reduction is applied below this threshold level
RATIO	The relationship between how much the input signal is below the threshold and how much the output signal is below the threshold
KNEE	When active, the Ratio gradually increases around the Threshold, starting from 1 : 1.1 and gradually reaching the user-selected Ratio, rather than being applied abruptly at the Threshold level. Higher values result in a wider transition range.
ATTACK	Time it takes for the expander to restore approximately two-thirds of the applied gain Expander becomes active when the input signal either: 1. Exceeds the threshold 2. Is still below the threshold and increases, reducing the targeted amount of gain reduction
HOLD	Amount of time in ms until the release phase is engaged
RELEASE	Defines time taken to apply approximately two-thirds of the targeted gain reduction. Becomes active when the input signal either: 1. Falls below the threshold 2. Is already below the threshold and decreases, increasing the new targeted gain reduction

WING Gate/Expander

A combination of a noise gate and downward expander. This effect is available in the GATE processing slot. The processor applies either an immediate, fixed gain reduction (noise gate) or a proportional gain reduction (expander).

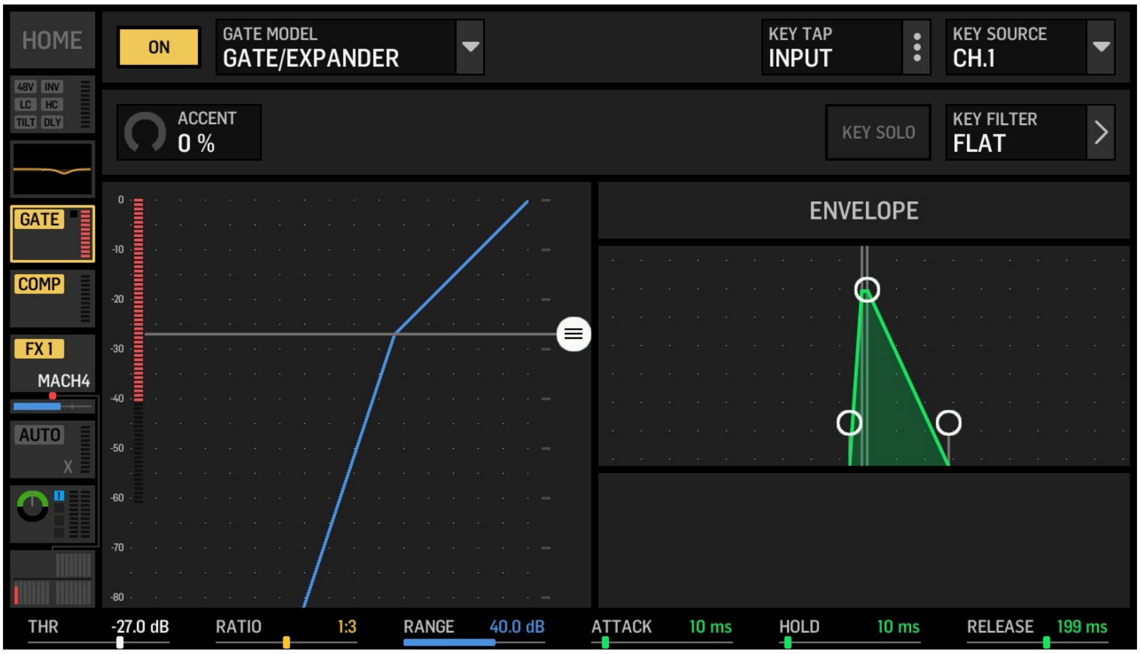

Figure 94: WING Gate/Expander

Parameter	Function
ACCENT	Briefly boosts the gain when the gate/expander opens, helping to restore transients that may have been attenuated by the gate/expander.
THR	Gain reduction is applied below this threshold level
RATIO	The relationship between how much the input signal is below the threshold and how much the output signal is below the threshold
RANGE	Defines the maximum amount of gain reduction the processor will apply.
ATTACK	Time it takes for the expander to restore approximately two-thirds of the applied gain reduction Gate/Expander becomes active when the input signal either: 1. Exceeds the threshold 2. Is still below the threshold and increases, reducing the targeted amount of gain reduction
HOLD	Amount of time in ms until the release phase is engaged

Parameter	Function
RELEASE	Defines time taken to apply approximately two-thirds of the targeted gain reduction. Becomes active when the input signal either: 1. Falls below the threshold 2. Is already below the threshold and decreases, increasing the new targeted gain reduction

Even 88 Comp

Based on the compressor section of the same British console series introduced in the early 2000s that inspires the Even 88-Formant and Even 88 Gate. This compressor is renowned for its clean and smooth dynamics control and detailed sound that doesn't get muddy, even when pushed hard. It features sophisticated double time constant auto Attack and triple time constant auto Release circuits.

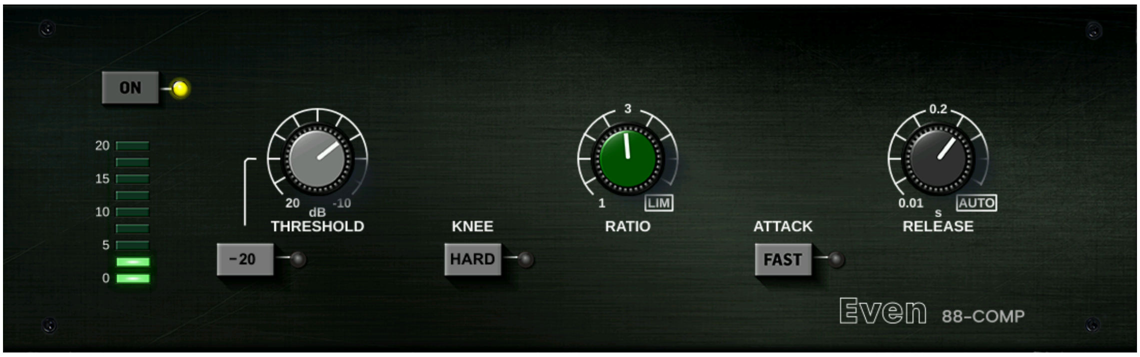

Figure 95: Even 88 Comp

Parameter	Function
THRESHOLD	Threshold in dB, when reached, gain reduction is applied
-20	Lowers by 20dB the value set by THRESHOLD
HARD	Switches between a soft (when disengaged) and hard knee (when engaged) compression curve
RATIO	The relationship between the input signal's excess over the threshold and the output signal's excess over the threshold
FAST	Toggles between automatic Attack time ranges: 1.5–5 ms when disengaged, and 0.1–5 ms when engaged
RELEASE	Defines time taken to restore approximately two-thirds of the applied gain reduction Becomes active when the input signal either: 1. Falls below the threshold, restoring gain toward unity 2. Decreases while still above the threshold, reducing the targeted amount of gain reduction

One Knob Compressor

Probably the most transparent compressor on WING, this effect can achieve large amounts of gain reduction without the pumping artifacts. The Attack and Release times are adjusted automatically in response to the Crest Factor, the difference between the peak and RMS audio levels.

The Auto Gain function automatically adjusts the output gain to compensate for the gain reduction applied by the compressor.

Figure 96: One Knob Compressor

Listen Mic Toolkit Compressor

Based on the talkback microphone limiter from a British console, LMT can easily produce the large, explosive drum sound characteristic of the 1980s.

The effect consists of a transient shaper (the three knobs on the left) and a compressor section (the two knobs on the right).

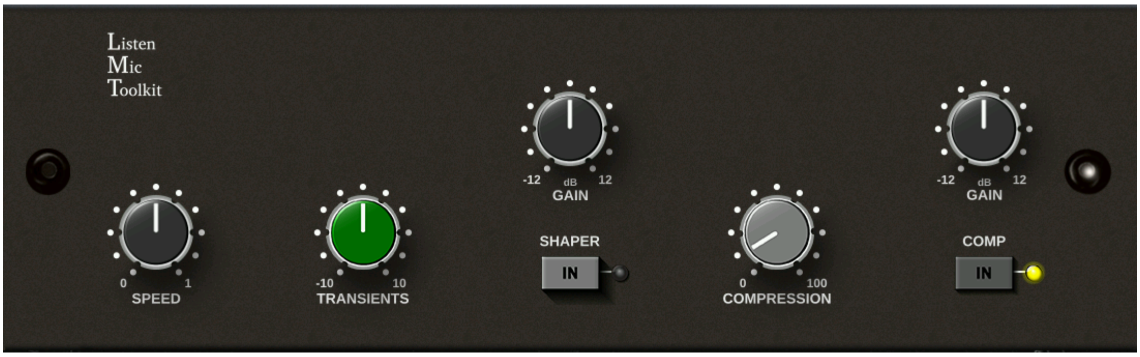

Figure 97: Listen Mic Toolkit Compressor

Parameter	Function
SPEED	Adjusts the reaction time of the transient shaper
TRANSIENTS	Increases or decreases the gain reduction during the Attack phase. Note that the transient shaper (except T GAIN, which is independent) is only active when the compressor is on
T GAIN	Adjusts the output gain of the transient shaper

Parameter	Function
SHAPER IN	Turns the transient shaper on or off
COMPRESSION	Adjusts the gain reduction applied by the compressor
C GAIN	Adjusts the output gain of the compressor
COMP IN	Turns the compressor on or off

BDX 160 Comp

A compressor based on the first solid-state VCA compressor, introduced in the 1970s. Its sound is often described as “aggressive” and “in-your-face,” and is widely used in base, guitars and vocals.

Figure 98: BDX 160 COMP

Parameter	Function
THRESHOLD	Threshold when reached, gain reduction is applied. Two LEDs indicate whether the signal is below or above the threshold.
COMPRESSION	Adjusts the ratio between how much the input signal is below the threshold and how much the output signal is below the threshold
OUTPUT GAIN	Gain compensation for changes introduced by the compression process

BDX 560 Easy

Multiple variations were created following the release of the first solid-state VCA compressor. BDX 560 Easy recreates one of these later designs, which incorporated a soft knee compression curve, as well as “beyond infinity” compression ratios.

Figure 99: BDX 560 Easy

Parameter	Function
EASY	Enables a soft knee compression curve , resulting in a gradual ratio increase around the threshold until reaching the full COMPRESSION RATIO
THRESHOLD	Threshold when reached, gain reduction is applied. Two LEDs indicate whether the signal is below or above the threshold.
COMPRESSION RATIO	Adjusts the compression ratio. From 2:1 to 10:1, functions as a typical compression ratio. At ∞:1, the compressor acts as a limiter. When set to negative values beyond this, the output signal becomes quieter as the input exceeds the threshold. For example, at -4:1, for every 4 dB above the threshold, the output is attenuated to 1 dB below the threshold.
OUTPUT GAIN	Gain compensation for changes introduced by the compression process

DRAW MORE Comp

DRAW MORE Comp is based on a sophisticated rack-mounted compressor/limiter. The compressor section features a soft knee that gradually increases the ratio, reaching its full value for signals 10 dB above the threshold.

An additional peak limiter follows the compressor, effectively handling any remaining transients. AUTO mode dynamically adjusts the Attack and Release times based on the incoming signal, delivering more transparent results on unpredictable sources or full mixes.

All other parameters function as described in the WING Compressor section.

Figure 100: DRAW MORE Comp

RED3 Compressor

The RED3 Compressor is based on a British VCA compressor known for its highly transparent sound, making it a popular choice for mastering applications.

All other parameters are defined as described in the WING Compressor section.

Figure 101: RED3 Compressor

SOUL 9000

Based on the compressor section of a British console, SOUL 9000 has an aggressive character often associated with modern records. Its attack time is program-dependent and has a soft knee curve.

Figure 102: SOUL 9000

Parameter	Function
THR	Threshold in dB, when reached, gain reduction is applied
RELEASE	Defines time taken to restore approximately two-thirds of the applied gain reduction. Becomes active when the input signal either: 1. Falls below the threshold, restoring gain toward unity 2. Decreases while still above the threshold, reducing the targeted amount of gain reduction
RATIO	The relationship between the input signal's excess over the threshold and the output signal's excess over the threshold
FAST	Switches from a program-dependent Attack (ranging from 3 ms to 30 ms). A fast Attack time of 3 ms is used to achieve 20 dB of gain reduction.
PEAK	Activating this setting changes the compression curve from soft knee to hard knee.

SOUL Bus Comp

Designed similarly to the British bus compressor known to "glue" mixes together. The AUTO release setting has two stages, resulting in higher gain reduction values being restored quickly, while lower gain reduction values are restored more slowly. This compressor is ideal for drums, subgroups and the main mix.

All other parameters are defined as described in the WING Compressor section.

Figure 103: SOUL Bus Comp

Even Comp/Lim

Based on a diode-bridge compressor and limiter introduced in the 1980s, this unit is commonly used on subgroups, drums and main mix. Both the compressor and limiter sections feature two automatic recovery (release) modes. When Auto is selected, recovery times adapts dynamically; the greater the amount of gain reduction, the longer the recovery time.

Even Comp/Lim features an additional slow attack option, not available in the original hardware.

Figure 104: Even Comp/Lim

Parameter	Function
COMPRESS RECOVERY	auto 1 ranges from 40 ms to 800 ms auto 2 ranges from 150 ms to 1500 ms
COMPRESS ATTACK	Switches between fast (3 ms ± 1ms) and slow (6 ms ± 1ms) attack times
LIMIT RECOVERY	auto 1 ranges from 40 ms to 800 ms auto 2 ranges from 150 ms to 1500 ms
LIMIT ATTACK	Switches between fast (2 ms ± 1ms) and slow (4 ms ± 1ms) attack times

All other parameters function as described in the WING Compressor section.

Eternal Bliss

Based on a modern German compressor and can deliver anything from transparent, subtle compression to bold and creative dynamic shaping.

Similar to BDX560 Easy, this unit also supports negative ratio values, where the output signal becomes quieter as the input exceeds the threshold. Note that the negative ratio values on eternal bliss can exceed those of BDX560 Easy.

Figure 105: Eternal Bliss

Parameter	Function
Auto Fast	Automatically shortens attack time when loud transients are detected, then returns it to the original setting afterward
Anti Log	Changes release curve from linear to exponential, where gain reduction is restored slowly at first, then accelerates until returning to unity gain. This also increases the likelihood of pumping artifacts
Gr Limit	Sets maximum gain reduction the compressor can apply, regardless of other settings
On	Activates Gr Limit
GRL	LED illuminates when the Gain Reduction Limit is actively constraining the gain reduction
Ratio	Adjusts the compression ratio. When set to negative values, output signal becomes quieter as input exceeds the threshold. For example, at -4, for every 1 dB above the threshold, the output is attenuated to 4 dB below the threshold

76 Limiter Amp

76 Limiter Amp is based on a legendary FET compressor introduced in the late 1960s, renowned for its versatility. 76 Limiter Amp can deliver everything from transparent compression to intentionally distorted and aggressive tones ideal for bass, guitar and vocals.

Its gritty and distorted character is made possible by its notably fast attack and release times, which can interact with waveform itself. Click on ALL to engage all ratio buttons, a well-known trick on the original hardware that intensifies the compression.

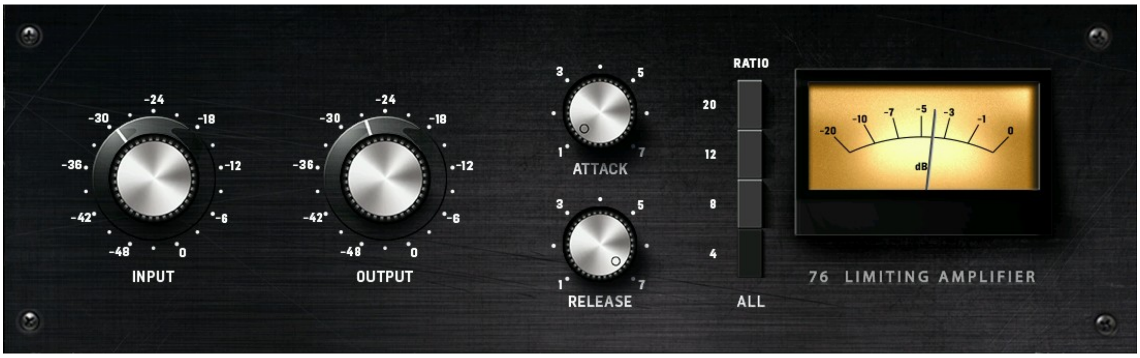

Figure 106: 76 Limiter Amp

Parameter	Function
INPUT	Adjusts both input gain and threshold simultaneously. Increasing this value causes more of the signal to exceed the threshold, resulting in greater gain reduction

Parameter	Function
OUTPUT	Adjusts output gain of compressor to compensate for the level changes caused by the INPUT control and the applied gain reduction
ATTACK	Adjusts attack time from 20 µs (0.02 ms) at the fully counterclockwise position to 80 µs (0.08 ms) at the fully clockwise position
RELEASE	Adjusts release time from 50 µs (0.05 ms) at the fully clockwise position to 1.1 s at the fully counterclockwise position
RATIO	Adjusts compression ratio. Engaging the "ALL" mode activates all ratio buttons simultaneously, resulting in a more aggressive compression effect

LA Leveler

LA Leveler is based on one of the most iconic optical compressors introduced in the early 1960s, renowned for its smooth and natural vocal compression characteristics.

In the original hardware, gain reduction was controlled by an optical cell comprising a light-emitting panel and a photoresistor. The input signal determined the intensity of the emitted light, and the photoresistor responded accordingly, controlling the amount of gain reduction applied. This opto cell behavior is naturally program-dependent and frequency-sensitive.

The release phase consists of multiple stages, initially restoring most of the gain reduction quickly, then slowing down progressively over time.

Figure 107: LA Leveler

Parameter	Function
GAIN	Controls post-compression gain to compensate for attenuation introduced by the compression. This GAIN control is separate from the standard GAIN control at the top of the GUI. The 0-100 scale is relative and does not represent absolute dB values

Parameter	Function
PEAK REDUCTION	Adjusts threshold. Turning the control to higher values lowers the threshold, making the compressor react to quieter signals. The 0-100 scale does not correspond to dB values
LIMIT/COMPRESS	Switches between a compression ratio of approximately 3:1 (COMPRESS) and an infinite ratio (LIMIT)

Fair Kid

Fair Kid is based on the legendary hardware known for being the heaviest compressor ever built, weighing over 30 kg due to its 20 vacuum tubes, 14 transformers, and robust metal chassis. It is commonly used on vocals, drums, pianos, and mix buses.

This compressor used vacuum tubes as attenuators. As the input signal increases, the current to the tubes' grid decreases, resulting in more gain reduction, approaching a limiting ratio. This results in a variable compression ratio, which is why these compressors are known as variable-mu types ("mu" or "µ" refers to gain).

Figure 108: Fair Kid

Parameter	Function
OUT GAIN	Same parameter as the GAIN control at the top of the GUI
INPUT GAIN	Controls attenuation applied to the input signal before it reaches the tube stage. Increasing this value can introduce more harmonic distortion, which may be desirable for coloration. Unity gain is maintained at the default setting of -10 dB
THRESHOLD	Level above which the compressor applies gain reduction. Adjusts the compressor's threshold. Higher values correspond to lower thresholds. The 0–10 scale is relative and does not represent absolute dB values

Parameter	Function
TIME CONSTANT	6-position switch moves between various attack and release times: • Position 1: Attack 200 µs; Release 300 ms • Position 2: Attack 200 µs; Release 800 ms • Position 3: Attack 400 µs; Release 2 s • Position 4: Attack 800 µs; Release 5 s • Position 5: Attack 200 µs; Release 2 s for individual transients and 10 s for lower peaks • Position 6: Attack 400 µs; Release 300 ms for transients
DC BIAS	Lowering this value from its default at 1.00 (Factory Cal) broadens the compression knee. This effect can be so wide that it acts just as a reduction in the initial compression ratio of 2:1

No-Stressor

The hardware that inspired the No-Stressor was itself influenced by both ultra-fast FET compressors and the smooth optical compressors of the 1960s. It is known for its versatility, offering everything from transparent compression to heavily crushed, aggressive sounds. This effect can be used anywhere; from drums, bass, guitar and vocals.

The scale for the 4 knobs on the compressor is relative and does not correspond to absolute dB values.

Figure 109: No-Stressor

Parameter	Function
INPUT	Adjusts input signal's gain and the threshold
ATTACK	Adjusts attack time from 50 µs to 30 ms
RELEASE	Adjusts release time from 50 ms to 3.5 s. When the ratio is set to 10:1, the release time can reach up to 20 s
OUTPUT	Controls post-compression gain to compensate for attenuation introduced by the compression. This GAIN control is separate from the standard GAIN control at the top of the GUI

Parameter	Function
RATIO	Adjusts compression ratio. When set to 10:1, the compressor emulates the attack and release times of optical compressors. All settings except 20:1 and NUKE have a soft knee

PIA 2250 Rack

PIA 2250 Rack is based on a 500-series American-made compressor, renowned for its ability to operate in either feed-back or feed-forward detection modes. This compressor is ideal for individual drum mics, bass and guitars.

Figure 110: PIA 2250 Rack

Parameter	Function
THRESHOLD	Threshold where the compressor applies gain reduction
RATIO	Adjusts compression ratio
ATTACK	Adjusts program-dependent attack time
RELEASE	Adjusts the release time from 50 ms at the fully counterclockwise position to 3 s at the fully clockwise position
KNEE	Switches between a soft and hard knee compression curve
TYPE	Switches between OLD (feed-back detection) and NEW (feed-forward detection) modes. In OLD mode, the sidechain signal controlling the compressor is taken after the gain reduction stage, producing smoother compression. In NEW mode, the sidechain signal is taken before the gain reduction stage, resulting in more transparent and precise compression

LTA100 Leveler

LTA100 Leveler is based on an American compressor that combines tube and solid-state components.

Figure 111: LTA100 Leveler

Parameter	Function
ATTACK	3-position switch to adjust the attack time
RELEASE	3-position switch to adjust the release time. Note that the release is also signal-dependent: greater amounts of gain reduction result in longer release times
GAIN	Controls post-compression gain to compensate for attenuation introduced by the compression. This GAIN control is separate from the standard GAIN control at the top of the GUI
GAIN REDUCTION	Adjusts threshold. Turning the control to higher values lowers the threshold, making the compressor react to quieter signals. The 0-10 scale is relative and does not correspond to dB values Note that the ratio also increases as the gain reduction increases

PSE/LA COMBO

This effect combines fully functional versions of the PSE-545 SOURCE EXTRACTOR and the LA LEVELER. Their respective controls are explained in their dedicated sections.

Figure 112: PSE/LA COMBO

Combinator

C5-COMBINATOR is a 5-band compressor with variable crossover slope and minimal phase shifting, inspired by the Behringer Combinator MDX-8000. The controls enclosed by the colored rectangle are specific to each filter band. To select a different band, click on the desired filter in the graphical representation on the left. The rectangle on the right changes color to indicate which band is currently selected for editing.

Figure 113: C5-Combinator

Parameter	Function
Attack	Adjusts attack time in a relative scale of 0-20
Release	Adjusts release time from 30 ms to 3000 ms
Auto	Enables auto release time
SBC (switch)	When enabled, Spectrum Balance Control automatically adjusts make-up gain for each band to maintain a relatively uniform level across all five bands
SBC (knob)	Adjusts how fast the automatic make-up gain is applied. The scale does not correspond to absolute time values
Thresh (top row)	Adjusts threshold for all five bands
Gain	Controls post-compression gain for all five bands to compensate for attenuation introduced by the compression
Ratio	Adjusts compression ratio for all five bands
Thresh (bottom row)	Adjusts threshold for each individual band, relative to the global threshold set with the main Threshold knob. For example, if the main threshold is set to -10 dB and the per-band threshold control is set to -4 dB, the resulting threshold for that band is -14 dB
Gain	Controls post-compression gain for all each individual band to compensate for attenuation introduced by the compression
Bypass	Disables compression on individual bands
Solo	Enables solo on the selected band
Width	Adjust frequency range of the selected band

Parameter	Function
Slope	Switches slope of the band-pass filters between 12 dB and 48 dB per octave
Mix	Dry/wet control

Precision Limiter

Provides precise peak control to prevent digital clipping and distortion.

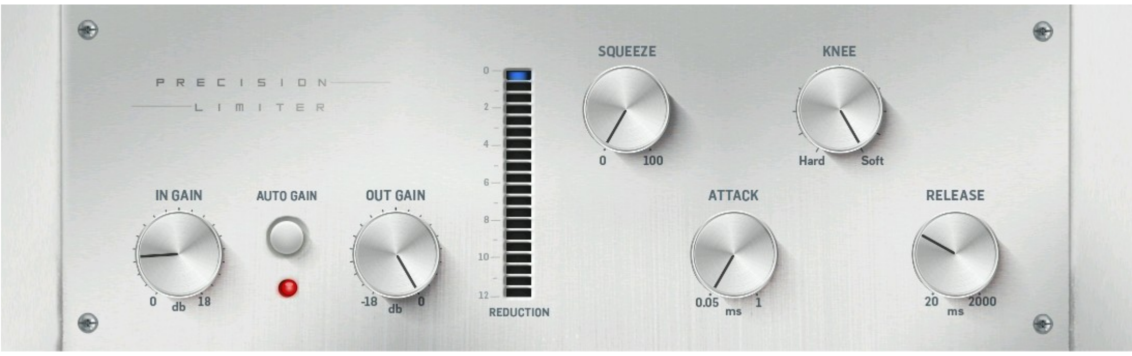

Figure 114: Precision Limiter

Parameter	Function
IN GAIN	Adjusts gain applied to the input signal
AUTO GAIN	Enables an additional long-term gain detection, allowing automatic gain control for signals with a wide dynamic range
OUT GAIN	Adjusts gain applied to the output signal after limiting has been applied
SQUEEZE	Increases perceived loudness. Higher settings may introduce distortion artifacts
KNEE	Gradually transitions compression curve from a completely hard knee (0) to a wide soft knee (10)
ATTACK	Adjusts attack time from 50 μs to 1 ms
RELEASE	Adjusts release time from 20 ms to 2 s

Wave Designer

Inspired by a German transient designer, this effect allows independent control over a signal's attack and sustain. Ideal for enhancing the punch and presence of drums, percussion, guitars and other sources.

Figure 115: Wave Designer

Auto Rider

AUTO RIDER automates gain adjustments of the selected channel to maintain a relatively consistent perceived volume.

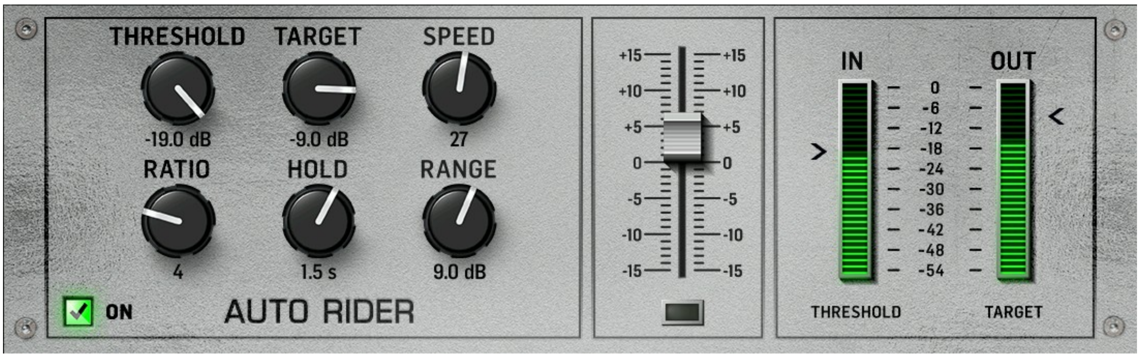

Figure 116: Auto Rider

Parameter	Function
THRESHOLD	Sets threshold the input signal must exceed for the effect to activate
TARGET	Sets target output level that AUTO RIDER should maintain
SPEED	Adjusts speed at which the fader moves
RATIO	The gain adjustments are applied following a logarithmic curve. This means that as the RATIO increases, the gain changes become larger at first but then diminish, approaching the limit defined by the RANGE
HOLD	Adjusts how long the fader remains at its adjusted position before gradually returning to unity gain once the input signal falls below the threshold.
RANGE	Sets maximum gain change (either attenuation or amplification) that the processor is allowed to apply to the signal

Ducker

A ducker attenuates the signal level by a fixed amount when it exceeds the set threshold. This processor is commonly used to create the "radio host effect," where background music is automatically attenuated when the host begins speaking.

To achieve this, the ducker is inserted on the music channel, and the host's microphone is selected as the KEY SOURCE.

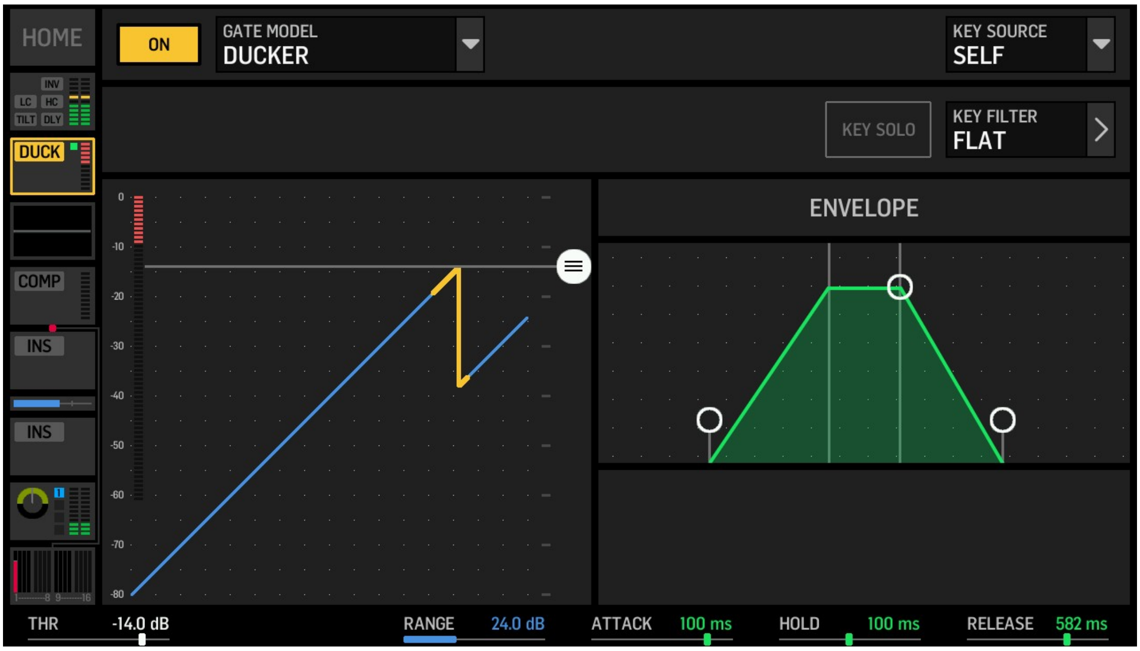

Figure 117: Ducker

Parameter	Function
THR	Threshold where the ducker becomes active and attenuates the signal
RANGE	Adjusts attenuation applied to the signal
ATTACK	Defines the time it takes for the ducker to apply approximately two thirds of the target attenuation
HOLD	Amount of time ducker takes to the start of the release phase
RELEASE	Defines the time it takes for the compressor to restore approximately two-thirds of the applied attenuation

SOUL 9000 Gate

Figure 118: Ducker

Parameter	Function
RELEASE	Defines time taken to apply approximately two-thirds of the targeted gain reduction. Becomes active when the input signal either: 1. Falls below the threshold 2. Is already below the threshold and decreases, increasing the new targeted gain reduction
THRESHOLD	Gain reduction is applied below this threshold level
2:1	Switches between a noise gate with fixed attenuation and a downward expander with 2:1 ratio
RANGE	Defines the maximum amount of gain reduction the processor will apply
HOLD	Amount of time in ms until the release phase is engaged
FAST	When enabled, Attack time is set to 0.1 ms. When disabled, default Attack time is 1.5 ms

Even 88 Gate

Figure 119: Even 88 Gate

Parameter	Function
FAST	Toggles between an attack time of 500 µs (not engaged) and 50 µs (engaged)
HYSTERESIS	Offsets threshold for opening the gate by the amount specified, compared to the threshold for closing the gate. For example, if THRESHOLD is set to -30 dB and HYSTERESIS to 5 dB, the gate opens at -25 dB and closes at -30 dB
THRESHOLD	Gain reduction is applied below this threshold level
-40	Lowers the gate's threshold range by 40 dB when engaged. Threshold can be adjusted between -40 dB and 0 dB. When engaged, the available range shifts to between -80 dB and -40 dB
RANGE	Defines the maximum amount of gain reduction the processor will apply
RELEASE	Defines time taken to apply approximately two-thirds of the targeted gain reduction. Becomes active when the input signal either: 1. Falls below the threshold 2. Is already below the threshold and decreases, increasing the new targeted gain reduction

DRAW MORE 241

DRAW MORE 241 is modeled after the gate/expander section of the same analog unit that inspired the DRAW MORE COMP. It provides a streamlined gating and expanding effect. Ratio is dependent on the input signal level—acting as a gentle expander close to the threshold and transitioning to a more aggressive gate as the signal drops further below it. Two release time settings are available. Enabling SLOW release is common for sound sources with a longer decay or reverb tails.

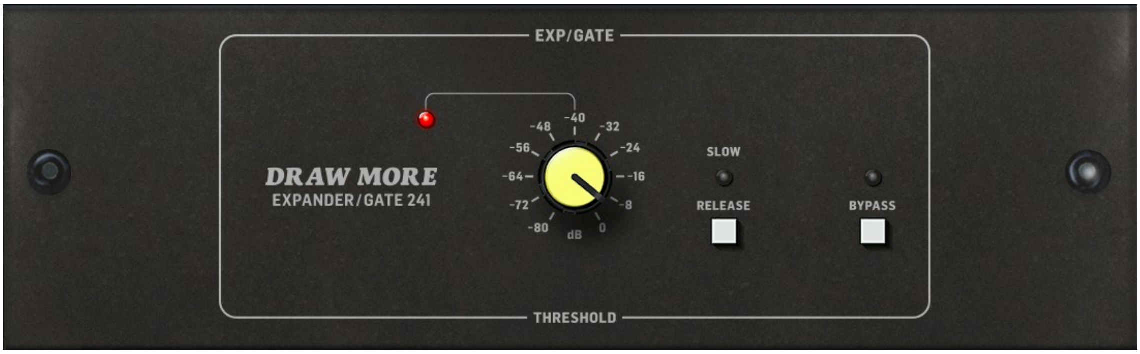

Figure 120: DRAW MORE 241

BDX902 De-Esser

Similar to the original analog hardware, BDX902 operates independently of the overall input signal level. The audio signal is divided into two bands. Instead of relying on a fixed threshold, it analyzes the level difference between the high-frequency band and the full-range audio material. It then attenuates the sibilant range accordingly. This approach allows it to remain effective in both loud and quiet sections of a vocal performance, without requiring manual adjustment of the threshold.

Figure 121: BDX902 De-Esser

Parameter	Function
SOLO	Solos high-frequency band to monitor and fine-tune only the sibilant portion of the signal
HF ONLY	When engaged, attenuation is applied exclusively to the high-frequency band, leaving the rest of the frequency spectrum unaffected. When disengaged, the attenuation is applied to the whole frequency spectrum
FREQUENCY	Determines crossover frequency where the signal is split into low and high-frequency bands for analysis
RANGE	Adjusts amount of attenuation in dB that is applied. A recommended range is indicated

2-Band De-Esser

Based on a German-made de-esser, this unit provides independent attenuation on two different frequency ranges, either in stereo or mid-side mode.

Note that 2-Band De-Esser is only available in the FX Rack and must be used as an insert point.

The four knobs have a relative scale of 1 to 50 and do not correspond to absolute dB values.

Figure 122: 2-Band De-Esser

Parameter	Function
LOW	Determines how much attenuation is applied by the lower-frequency de-essing filter. This filter targets the lower portion of the sibilant range
HIGH	Determines how much attenuation is applied by higher-frequency de-essing filter. This filter targets the higher portion of the sibilant range
FEMALE/MALE	Adjusts center frequency of the lower de-essing filter
M/S MODE	Switches from stereo to mid-side (M/S) mode. When in M/S mode, the left LOW and HIGH knobs control the attenuation on the MID channel
LOW (SIDE)	Determines how much attenuation is applied by lower-frequency de-essing filter on the SIDE channel
HIGH (SIDE)	Determines how much attenuation is applied by higher-frequency de-essing filter on the SIDE channel

Source Extractor

Based on a modern British noise gate, Source Extractor offers straightforward gating. THRESHOLD works as indicated in other gate effects. RANGE determines the attenuation applied by the noise gate.

Figure 123: Source Extractor

FAST and PEAK work together to modify the time constants. FAST toggles between fast and slow attack and release times. PEAK toggles between peak and RMS detection.

	RMS		PEAK	
	Attack (ms)	Release (ms)	Attack (ms)	Release (ms)
FAST	100	100	20	200
SLOW	200	200	20	1000

Table 11.33

11.3 Channel Strips

Channel strips combine multiple processing effects, such as EQ, compression, gating and more into a unified module. These strips are found in the FX Rack and are designed to be used as insert effects.

EVEN Channel

Combines Even 88 Gate, Even 88-Formant and Even Comp/Lim.

SOUL Channel

Combines SOUL 9000 Gate/Expander, SOUL Analog and SOUL 9000.

Vintage Channel

Combines 76 Limiter Amp, Pulsar EQ and LA Leveler.

Bus Channel

Combines SOUL Warmth, Even 84 and SOUL Bus Comp.

Mastering

Combines Tape Machine, Mach EQ, Ultra Enhancer and Precision Limiter.

11.4 Reverbs

WING includes a fine selection of reverbs including digital and analog studio staples, vintage mechanical reverb emulations and guitar pedals.

WING includes a plethora of reverbs; from digital, analog, vintage mechanical emulations and guitar pedals.

Algorithmic Reverbs

The first six reverb devices—Hall, Room, Chamber, Plate, Concert, and Ambience—are based on distinct algorithm banks from a renowned digital reverb unit. Although each reverb presents a unique sonic character, they all share the same graphical interface.

Early reflections in each reverb are emulated using various combinations of delay lines, which are then blended with the reverb tail:

- Pre-echoes: A feedback loop created by routing a signal copy from after the DIFFUSION stage back to before the DIFFUSION stage.
- Reflections: A delay line taken either before or after the DIFFUSION stage.
- Delay: A delay line with its own feedback loop, sourced either before or after DIFFUSION.
- Dry delay: A delay line taken directly from the effect input.

Most of the parameters affect the reverb tail, which is combined with the early reflections before being sent to the effect output.

Parameters are grouped across up to three layers, accessible via the navigation buttons in the lower right corner of the interface.

Each parameter is described only the first time it appears in the following sections.

Hall Reverb

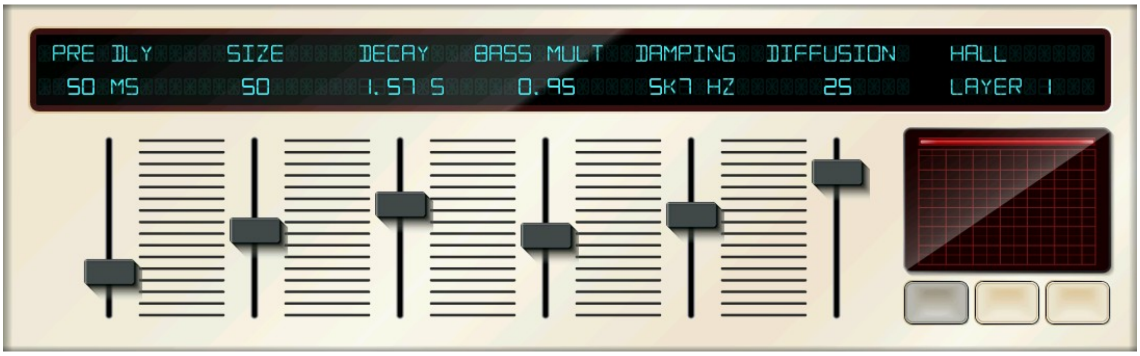

Figure 124: Hall Reverb

Parameter	Function
PRE DLY (Pre-Delay)	Defines delay, in milliseconds, between the dry signal and the onset of the reverb effect
SIZE	Roughly corresponds to the longest dimension of the acoustic space that is being recreated. SIZE serves as a master control of the DECAY time and SPREAD
DECAY	Specifies the time in ms for the mid-range frequencies of the reverb to decay by 60 dB from their initial level. This time is affected by the SIZE parameter
BASS MULT (Bass Multiply)	Defines what factor the DECAY time is multiplied by to determine the reverb duration for low frequencies. For instance, if BASS MULT is set to 2.00 and DECAY is 0.80 seconds, the resulting bass decay time will be 1.60 seconds
DAMPING	Defines cut-off frequency of a 6 dB per octave high-cut filter. This roll-off is applied both to the early reflections and to the reverb tail, contributing to a smoother and more natural-sounding reverberation
DIFFUSION	Controls how the early reflections are dispersed over time. Lower values produce fewer, louder reflections, simulating spaces with low acoustic diffusion. Higher values yield more, quieter reflections, emulating environments with greater acoustic diffusion that disperse energy over time
LO CUT	Sets cut-off frequency for the low-cut filter applied to the reverberation
HI CUT	Sets cut-off frequency for the high-cut filter applied to the reverberation

Parameter	Function
SHAPE	SHAPE and SPREAD define the contour and duration of the reverberation envelope. SHAPE controls the steepness of the reverb buildup. Lower SHAPE values result in a steep buildup. As the SHAPE value increases, the buildup becomes more gradual
SPREAD	SHAPE and SPREAD define the contour and duration of the reverberation envelope. SPREAD, in conjunction with SIZE, determines how long the envelope defined by SHAPE unfolds over time
MOD SPD	Modulates speed of the random variations in the reverb tail's delay times and pitch

Room Reverb

The Room Reverb bank introduces additional parameters, explained in the table below, which control the unit's pre-echo delay lines. For information on the remaining parameters, please refer to the Hall Reverb section.

Parameter	Function
SPIN	Randomly modulates delay times and pitch of the reverb tail, creating a more natural, less harsh, and more subtle reverb sound
ECHO L / ECHO R	Controls time delay applied to ECHO feedback loop
FEED L / FEED R	Controls amount of signal sent to the ECHO feedback loop

Chamber Reverb

Chamber Reverb provides the same parameters as Room Reverb. The parameters FEED L and R are replaced by ECHO L and ECHO R, which control the left and right output levels of the ECHO feedback loop before being summed with the reverb tail.

Plate Reverb

Plate Reverb includes some of the parameters described previously and introduces the Attack parameter, which controls how quickly the reverb develops following the input signal. Lower values result in a slower buildup, while higher values produce a faster onset. The influence of the Attack parameter is limited to a maximum of 50 milliseconds.

Concert Reverb

Concert Reverb emulates a digital concert hall reverb, includes some of the parameters described previously and introduces the following:

Parameter	Function
DEPTH	Defines amplitude of the reverb envelope, effectively adjusting the listener's perceived position within the space—from closer to the front to farther toward the back of the hall
CHORUS	Like SPIN, introduces random variations in the delay times
REFL L / REFL R	The first pair of REFL L / R faders controls the delay time applied to the early reflections
REFL L / REFL R	The second pair of REFL L / R faders controls the level of the early reflections before being added to the reverb tail

Ambience Reverb

Ambience Reverb includes some of the parameters described previously and introduces the following:

Parameter	Function
TAIL GAIN	Adjusts level of the reverb tail before being added to the dry signal
MOD SPD	Modulates the speed of the random variations in the reverb tail's delay times and pitch

VSS3 Reverb

VSS3 Reverb is a true stereo reverb based on TC Electronic's renowned algorithms, offering a wide range of parameters for creating highly customizable, natural-sounding reverbs.

A comprehensive list of presets is displayed on the right side. To load a preset, navigate through the list until the desired preset is highlighted by the arrow, then click LOAD.

The parameters are organized across multiple layers. The main display presents a selection of the most essential parameters from all layers. Click the arrow in the top right corner to access four additional layers: MAIN, ER (Early Reflections), TAIL, and MOD (Modulation).

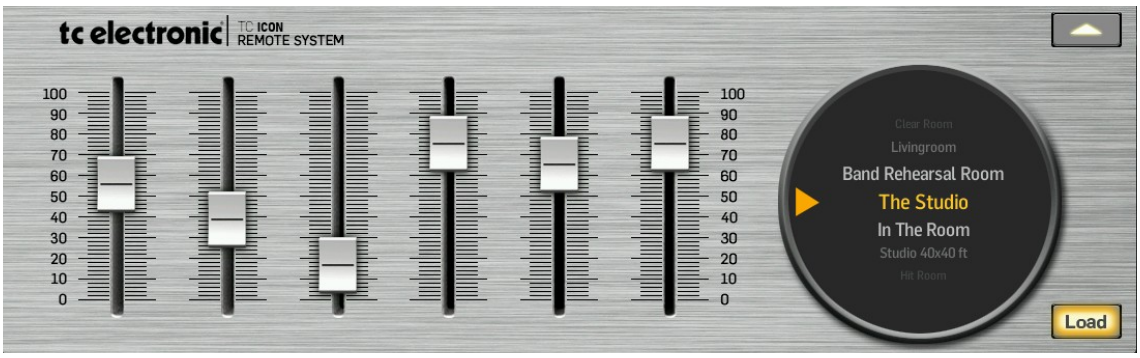

Figure 125: VSS3 Reverb

VSS3 home page parameters

Parameter	Function
DECAY	Determines overall duration of the reverb tail
LO DAMP	Sets attenuation value applied to the frequency range below the frequency set by LO SHV
HI DCY	Decay time for the high frequencies is determined by multiplying the master DECAY time by this factor
HI CUT	Sets cut-off frequency of the high-cut filter
REV LVL	Sets reverb TAIL level
ER LVL	Sets early reflections level

VSS3 MAIN page parameters

Parameter	Function
LO DCY / LOMID DCY / HIMID DCY / HI DCY	Sets the factor that the master DECAY is multiplied by and then applies this to each respective frequency band
LO X/O / MID X/O / HI X/O	LO X/O sets the crossover point between the LO and LOMID bands. MID X/O determines the crossover between the LOMID and HIMID bands. HI X/O sets the crossover between the HIMID and HI bands
DECAY	(same parameter as described above)
DIFFUSE	Controls how the early reflections are dispersed over time. Lower values produce fewer, louder reflections, simulating spaces with low acoustic diffusion. Higher values yield more, quieter reflections, emulating environments with greater acoustic diffusion that disperse energy over time
LO SHV	Sets the cut-off frequency where the attenuation defined by LO DAMP takes effect
LO DAMP	(same parameter as described above)
HI SOFT	Controls the intensity of a dedicated filter section associated with the HI CUT and HI DCY parameters, providing additional tonal shaping for the high-frequency band
HI CUT	(same parameter as described above)

VSS3 ER page parameters

Parameter	Function
ER TYPE	Selects acoustic space environment that the early reflections emulate
ER SIZE	Adjusts size of the acoustic environment that the early reflections emulate
ER POS	Adjusts perceived distance between the listener and the sound source by shaping the timing and intensity of the early reflections
ER PDLY	Sets delay between the direct sound and the onset of the early reflections
ER LCUT	Determines cut-off frequency of the low-cut filter applied to the early reflections
ER COL	Tone control for the early reflections
ER BAL	Adjusts stereo position of the early reflections
ER LVL	Adjusts overall level of the early reflections

VSS3 TAIL page parameters

Parameter	Function
REV TYPE	Selects different types of reverb tails
REV WID	Selects stereo width of the reverb tail
REV PDLY	Sets delay between the direct sound and the onset of the reverb tail
REV BAL	Adjusts stereo position of the reverb tail
REV LVL	Adjusts overall level of the reverb tail

VSS3 MOD page parameters

Parameter	Function
MOD TYPE	Selects modulation curve
MOD RATE	Sets modulation rate
MOD WID	Sets modulation depth

Vintage Room

Vintage Room provides natural sounding reverb inspired by the earliest digital reverberators that became popular during the 1980s. It excels in recreating natural acoustic ambiences with a warm and dense characteristic without sounding particularly artificial.

Figure 126: Vintage Room

Parameter	Function
REV DELAY	Sets delay between the direct sound and the onset of the reverb tail
DECAY	Determines duration of the reverb tail
ROOM SIZE	Roughly corresponds to the longest dimension of the acoustic space that is being recreated
DENSITY	Controls the density of reflections within the reverb tail. Lower values produce a sparser texture, while higher values create a tighter, more continuous tail with closely packed reflections
LOW / HIGH MULTIPLY	Sets the factor that the master DECAY is multiplied by and then applies this to each respective frequency band
ER DELAY L / R	Sets delay between the direct sound and the onset of the early reflections
LOW CUT	Determines cut-off frequency of the low-cut filter applied to the reverb
HIGH CUT	Determines cut-off frequency of the high-cut filter applied to the reverb
ER LEVEL	Adjusts overall level of the early reflections relative to the direct sound
LEVEL	Adjusts overall level of the reverb
FREEZE	Holds reverb tail for as long as the function is active
ADD	Allows additional incoming signals to be included in the reflection pattern generated by the FREEZE function

Vintage Reverb

Vintage Reverb is inspired by the first digital reverb unit from the late 1970s in Germany. Its sound is often described as clear, open and lush.

Figure 127: Vintage Reverb

Parameter	Function
DECAY	Determines duration of the reverb tail
PREDELAY	Sets delay between the direct sound and the onset of the reverb tail
LO MULTI	Sets the factor the master DECAY is multiplied by and is then applied to the low frequency band
LO CUT	Determines cut-off frequency of the low-cut filter applied to the reverb
HI MULTI	Sets the factor the master DECAY is multiplied by and is then applied to the high frequency band
HI CUT	Determines cut-off frequency of the high-cut filter applied to the reverb
MODULATE	Sets amount of modulation applied to the reverb
OUTPUT	The original hardware was designed with two output pairs—front and rear—to support quadraphonic sound setups. Each pair has distinct sonic characteristics. The OUTPUT selector determines which of these pairs is used
TRANSFORMER	Enables emulation of the output transformer

Vintage Plate

Vintage Plate is inspired by the classic plate reverberator developed in the late 1950s in Germany and widely used in a renowned British recording facility.

Figure 128: Vintage Plate

Parameter	Function
FILTER	Determines cut-off frequency of the low-cut filter applied to the reverb
DECAY	Determines duration of the reverb tail
PRE DELAY	Sets delay between the direct sound and the onset of the reverb tail
COLOUR	Adjusts overall tone of the reverb

Blue Plate

Figure 129: Blue Plate

Parameter	Function
PREDELAY	Sets delay between the direct sound and the onset of the reverb tail
SIZE	Roughly corresponds to the longest dimension of the acoustic space that is being recreated
DECAY	Determines duration of the reverb tail

Parameter	Function
DIFF	Controls how the early reflections are dispersed over time. Lower values produce fewer, louder reflections, simulating spaces with low acoustic diffusion. Higher values yield more, quieter reflections, emulating environments with greater acoustic diffusion that disperse energy over time
DAMPING	Defines cut-off frequency of a subtle high-cut filter
BASSMULT	Sets factor the master DECAY is multiplied by and then applies this to the low frequency band
XOVER	Sets cut-off frequency where the DECAY modification set by BASSMULT takes effect
LOCUT	Determines cut-off frequency of the low-cut filter applied to the reverb
HICUT	Determines cut-off frequency of the high-cut filter applied to the reverb
MOD	Adjusts modulation depth
SPD	Adjusts modulation speed

Gated Reverb

GATED REVERB emulates the classic gated reverb technique that gained popularity in the 1980s. This effect was originally created by placing a noise gate after a reverb, causing the reverb tail to be abruptly cut off once the signal dropped below a certain threshold.

Figure 130: Gated Reverb

Parameter	Function
PRE DELAY	Sets delay between the direct sound and the onset of the reverb tail
DECAY	Determines duration of the reverb tail
ATTACK	Controls how quickly the reverb develops following the input signal. Lower values result in a slower buildup, while higher values produce a faster onset

Parameter	Function
DENSITY	Controls the density of reflections within the reverb tail. Lower values produce a sparser texture, while higher values create a tighter, more continuous tail with closely packed reflections
SPREAD	Determines for how long the reverb's envelope unfolds
DIFFUSE	Controls how the early reflections are dispersed over time. Lower values produce fewer, louder reflections, simulating spaces with low acoustic diffusion. Higher values yield more, quieter reflections, emulating environments with greater acoustic diffusion that disperse energy over time
LO CUT	Determines cut-off frequency of the low-cut filter applied to the reverb
FREQ	Sets cut-off frequency of the high shelving filter
GAIN	Sets attenuation applied by the high shelving filter

Reverse Reverb

Figure 131: Reverse Reverb

Parameter	Function
PRE DELAY	Sets delay between the direct sound and the onset of the reverb tail
DECAY	Determines duration of the reverb tail
RISE	Adjusts rise length of the reverb's envelope
SPREAD	Determines for how long the reverb's envelope unfolds

Parameter	Function
DIFFUSE	Controls how the early reflections are dispersed over time. Lower values produce fewer, louder reflections, simulating spaces with low acoustic diffusion. Higher values yield more, quieter reflections, emulating environments with greater acoustic diffusion that disperse energy over time
LO CUT	Determines cut-off frequency of the low-cut filter applied to the reverb
FREQ	Sets cut-off frequency of the high shelving filter
GAIN	Sets attenuation applied by the high shelving filter

Delay/Reverb

Delay/Reverb combines two effect processors in one single slot. The left panel provides controls for the delay, while the right panel controls the reverb effect. Both the dry signal and the delay's output signal can be fed into the reverb.

Figure 132: Delay/Reverb

Parameter	Function
TIME	Adjusts delay time
FEEDBACK	Adjusts delay feedback
HICUT	Sets cut-off frequency of the low-cut filter in the feedback loop
LEVEL	Adjusts delay level relative to the dry signal
DELAY	Adjusts level of the delay output sent to the reverb input
DIRECT	Adjusts level of the dry signal sent to the reverb input
DECAY	Determines duration of the reverb tail
SIZE	Roughly corresponds to the longest dimension of the acoustic space that is being recreated

Parameter	Function
PREDEL	Sets delay between the direct sound and the onset of the reverb tail
DAMP	Defines cut-off frequency of a subtle high-cut filter
LOCUT	Determines cut-off frequency of the low-cut filter applied to the reverb

Shimmer Reverb

Shimmer Reverb emulates a combination of guitar pedals to achieve the "shimmer reverb effect", popular amongst guitarists.

Figure 133: Shimmer Reverb

Parameter	Function
SHIMMER	Adjusts the number of harmonics added to the signal
SHINE	Adjusts the progressive attenuation of each successive harmonic added by the SHINE parameter
PRE DELAY	Sets the delay between the direct sound and the onset of the reverb tail
DECAY	Determines the duration of the reverb tail
SIZE	Roughly corresponds to the longest dimension of the acoustic space that is being recreated
DAMPING	Defines the cutoff frequency of a subtle high-cut filter
LO CUT	Determines the cutoff frequency of the low-cut filter
HI CUT	Determines the cutoff frequency of the high-cut filter
RESET	Resets both cutoff frequencies

Spring Reverb

Inspired by a mechanical spring reverberator from the 1970s, Spring Reverb is often used on electric guitars.

Figure 134: Spring Reverb

Parameter	Function
DECAY	Determines the duration of the reverb tail
DENSITY	Controls the density of reflections within the reverb tail. Lower values produce a sparser texture, while higher values create a tighter, more continuous tail with closely packed reflections
BASS	Adjusts the level of the low frequencies
TREBLE	Adjusts the level of the high frequencies

11.5 Delays

Stereo Delay

Figure 135: Stereo Delay

Parameter	Function
TIME	Determines the delay time
FEEDBACK	Adjusts the amount of feedback
STEREO	The feedback loop for both stereo channels are independent of each other
CROSS	The left channel's feedback loop is fed back to the right channel and vice versa

Parameter	Function
MONO	The left and right feedback loops are summed and then fed back to both channels
FACTOR	Sets the factor the TIME parameter is multiplied by
PATTERN	Modifies delay time by a factor specific to the left and right channels
OFFSET	Adjusts time difference between the left and right channels
FB LOCUT	Determines cut-off frequency of the low-cut filter in the feedback loop
FB HICUT	Determines cut-off frequency of the high-cut filter in the feedback loop
LOW CUT	Determines cut-off frequency of the low-cut filter applied to the input signal
HIGH CUT	Determines cut-off frequency of the high-cut filter applied to the input signal

Ultratap Delay

Ultratap Delay provides flexible and straightforward control over each sound repetition for creative effects.

Figure 136: Ultratap Delay

Parameter	Function
WIDTH	Adjusts total stereo width of the repetitions
DIFFUSION	Emulates sound of early reflections instead of the direct sound
LO CUT	Determines cut-off frequency of the low-cut filter applied to the input signal
HI CUT	Determines cut-off frequency of the high-cut filter applied to the input signal
TIME	Determines delay time
FACTOR	Sets factor the TIME parameter is multiplied by

Parameter	Function
PRE DELAY	Sets delay between the direct sound and the onset of the repetitions
REPEATS	Sets the total number of repetitions
MOVE/JUMP/FOCUS/SPREAD	• MOVE: the panning moves in a linear way from one side to the other • JUMP: the panning alternates from hard left to hard right or vice versa • FOCUS: the repetitions start from hard left or right panning and alternate while the panning is progressively reduced until reaching the center • SPREAD: opposite of focus. The repetitions start panned center and alternate left and right while the panning progressively widens until reaching hard left and right. Note that there must be three or more repetitions for a stereo effect to be achieved. The first two repetitions in SPREAD mode will always be panned center
SLOPE	Adjusts progressive attenuation or amplification applied to the repetitions

Tape Delay

Tape Delay is modeled after a classic mid-1970s tape delay unit, which recorded the input signal onto magnetic tape and played it back to produce delayed repetitions. Changing the tape speed altered the delay time. The tape medium imparted a distinctive warmth and midrange-rich tone, making it a popular choice for slapback delays on vocals and electric guitars.

Figure 137: Tape Delay

Parameter	Function
TIME	Determines delay time
SUSTAIN	Adjusts amount of feedback
DRIVE	Adds harmonic distortion to the signal
FLUTTER	Emulates characteristic pitch modulation effect caused by variations in the tape speed

Oilcan Delay

Oilcan Delay is modeled after a vintage delay unit that used a fluid-filled can and a rotating pickup to create the delay effect. This design offered a more durable alternative to tape delays, which were susceptible to mechanical wear.

Figure 138: Oilcan Delay

Parameter	Function
TIME	Determines delay time
SUSTAIN	Adjusts amount of feedback
WOBBLE	Adjusts pitch modulation
TONE	Adjusts overall tone of the delay

BBD Delay

BBD DELAY is named after the “Bucket Brigade Device”. The term refers to the way the signal is passed through a chain of capacitors, similar to how firefighters pass buckets of water along a line. Each capacitor introduces a slight delay as it charges and discharges, cumulatively creating the delay effect. BBD DELAY emulates the warm, dark tone of these analog units, including the characteristic low-pass filtering used to reduce noise generated by the many capacitors in the circuit

Figure 139: BBD Delay

Parameter	Function
FEEDBACK	Adjusts amount of feedback
DELAY	Determines delay time

11.6 Modulation

Dimension CRS

Based on a late 1970s analog chorus, Dimension CRS introduces subtle modulation to enhance the stereo image and texture.

Figure 140: Dimension CRS

Parameter	Function
MODE	Toggles between mono or stereo input mode. When the LED is off, stereo inputs are summed to mono.
MIX	Adds DRY (unprocessed) signal to the effect's output
OFF, 1 – 4	Enables different levels of modulation, ranging from 1 to 4. Multiple modulation buttons can be activated simultaneously.
POWER	Bypass effect

Stereo Chorus

Stereo Chorus consists of three sections: the Chorus itself, a Modulator that controls the delay lines of the Chorus, and a set of filters.

Figure 141: Stereo Chorus

Parameter	Function
SPEED	Sets modulation rate in Hertz (Hz)

Parameter	Function
WAVE	Shapes modulation waveform from sine to triangle wave
SPREAD	Adjusts stereo width of the two chorus voices
PHASE	Adjusts phase offset between the two chorus voices
DELAY	Sets delay times for the left and right chorus voices independently
MIX	Adjusts balance between the processed (wet) and unprocessed (dry) signal
DEPTH	Adjusts how much the modulation affects each chorus voice
LO CUT	Determines cut-off frequency of the low-cut filter
HI CUT	Determines cut-off frequency of the high-cut filter
RESET	Resets both cut-off frequencies

Stereo Flanger

Stereo Flanger consists of three sections: the Flanger itself, the Feedback section for the Flanger effect, and a set of filters.

Figure 142: Stereo Flanger

Parameter	Function
FEED LC	Adjusts the cut-off frequency of the low-cut filter applied in the feedback loop
FEED HC	Adjusts the cut-off frequency of the low-cut filter applied in the feedback loop
FEEDBACK	Adjusts the amount of feedback
DELAY L/R	Sets the delay times for the left and right voices independently
RESET	Resets both cut-off frequencies
DEPTH L/R	Adjusts modulation intensity for the left and right voices independently

Parameter	Function
SPEED	Adjusts modulation rate in Hertz (Hz)
PHASE	Adjusts phase offset between the two chorus voices
LO CUT	Determines cut-off frequency of the low-cut filter
HI CUT	Determines cut-off frequency of the high-cut filter
RESET	Resets both cut-off frequencies

Rotary Speaker

Figure 143: Rotary Speaker

Parameter	Function
SLOW RATE	Adjusts slow rotation rate
FAST RATE	Adjusts fast rotation rate
FAST/SLOW/STOP	Toggles between fast rotation, slow rotation and stop positions
BASS	Adjusts transition rate of the bass speaker's rotation speed when changing between fast, slow, and stop positions
HORN	Adjusts transition rate of the horn speaker's rotation speed when changing between fast, slow, and stop positions
BALANCE	Adjusts mix between bass and horn speakers
DISTANCE	Emulates distance between the speaker and the listening position
MIX	Adjusts balance between the processed (wet) and unprocessed (dry) signal

Phaser

Figure 144: Phaser

Parameter	Function
MIX	Adjusts balance between the processed (wet) and unprocessed (dry) signal
RESONANCE	Accentuates the peak frequencies
STAGES	Determines number of all-pass filters in the phaser effect. More stages produce more peaks and throughs in the frequency response
RANGE	Sets lowest value that the phaser's rate can reach
DEPTH	Adjusts modulation amount applied by the LFO
ENV. MOD	Adjusts modulation amount applied the envelope
SPEED	Adjusts modulation rate
PHASE L/R	Adjusts phase offset between the left and right voices
SHAPE	Adjusts modulation waveform from triangle to sine and square waves
ATTACK	Sets attack time that modulates the rate
HOLD	Sets hold time that modulates the rate
RELEASE	Sets release time that modulates the rate

Tremolo/Panner

Figure 145: Tremolo/Panner

Parameter	Function
DEPTH (ENV MOD)	Adjusts how much the envelope modulates the tremolo DEPTH
SPEED (ENV MOD)	Adjusts how much the envelope modulates the tremolo SPEEDDEPTH
DEPTH	Adjusts modulation amount applied by the envelope
SPEED	Adjusts modulation rate
PHASE L/R	Adjusts phase offset between the left and right voices
SHAPE	Adjusts modulation waveform from triangle to sine and square waves
ATTACK	Sets attack time that modulates the tremolo depth
HOLD	Sets hold time that modulates the tremolo depth
RELEASE	Sets release time that modulates the tremolo depth

Mood Filter

Mood Filter emulates the filter section of a renowned analog synthesizer from the 1970s.

Figure 146: Mood Filter

Parameter	Function
DEPTH (ENV)	Adjusts modulation amount applied by the envelope
ATTACK	Sets attack time that modulates the cut-off frequency
HOLD	Sets hold time that modulates the cut-off frequency
RELEASE	Sets release time that modulates the cut-off frequency
BASE FREQ	Adjusts filter's cut-off frequency
TYPE	Selects filter type: low-pass, high-pass, band-pass, or notch
SLOPE	Toggles between a 12 and 24 dB per octave filter slope
RESONANCE	Adjusts filter's resonance
DRIVE	Adds saturation to the audio signal
MIX	Adjusts balance between the processed (wet) and unprocessed (dry) signal
DEPTH (LFO)	Adjusts modulation amount applied by the LFO
SPEED	Adjusts LFO's speed
PHASE	Adjusts phase offset between the left and right voices
WAVE	Selects LFO's waveform

Velvet Imager

Velvet Imager can be used to enhance or reduce the stereo content of an audio signal.

Figure 147: Velvet Imager

Parameter	Function
MS-WIDTH	Adjusts stereo width by modifying the ratio between the mid and side channels
STEREOIZE	Adjusts width of the artificial stereo impression

Parameter	Function
GAIN	Adjusts the effect's output gain
MODE	Toggles between two STEREOIZE operation modes: • K-Stereo: Based on a renowned mastering engineer's technique, it introduces subtle reflections to generate an artificial stereo image • Uses velvet noise to randomly shift the phase between the left and right channels
DEEP	When enabled, STEREOIZE effect is more pronounced

Tip: In Velvet mode with STEREOIZE set to 100%, the two output channels can be routed to separate speakers or speaker arrays to minimize phase cancellation effects.

Parameter	Function
DEPTH (ENV)	Adjusts modulation amount applied by the envelope
ATTACK	Sets attack time that modulates the cut-off frequency
HOLD	Sets hold time that modulates the cut-off frequency
RELEASE	Sets release time that modulates the cut-off frequency
BASE FREQ	Adjusts filter's cut-off frequency
TYPE	Selects filter type: low-pass, high-pass, band-pass, or notch
SLOPE	Toggles between a 12 and 24 dB per octave filter slope
RESONANCE	Adjusts filter's resonance
DRIVE	Adds saturation to the audio signal
MIX	Adjusts balance between the processed (wet) and unprocessed (dry) signal
DEPTH (LFO)	Adjusts modulation amount applied by the LFO
SPEED	Adjusts LFO's speed
PHASE	Adjusts phase offset between the left and right voices
WAVE	Selects LFO's waveform

11.7 Pitch

Stereo Pitch

Pitch shifting is commonly used in two ways. A subtle pitch shift added to the original signal creates a "doubler" effect, enhancing the timbre thickness. Alternatively, shifting the signal by an entire octave—up or down—can add a distinct layer to the mix.

Figure 148: Stereo Pitch

Parameter	Function
Mix	Adjusts balance between the processed (100%) and unprocessed (0%) signal
Pitch	Two knobs in the blue section adjust the pitch: one in semitones (±12) and the other in cents (±50)
Delay	Introduces a delay of up to 500 milliseconds
HiCut	Determines cut-off frequency of the high-cut filter
LoCut	Determines cut-off frequency of the low-cut filter

Dual Pitch

Dual Pitch features two independent sets of controls for channels A and B. Each set of controls is similar to the ones found in Stereo Pitch. Each channel also includes DELAY and PAN controls.

Figure 149: Dual Pitch

Parameter	Function
DELAY	Introduces a delay of up to 500 milliseconds to each channel separately
PAN	Adjusts pan value of each channel
SEMITONES	Independently adjusts pitch of channels A and B in semitones

Parameter	Function
CENTS	Independently adjusts pitch of channels A and B in cents
LEVEL	Adjusts level of channels A and B independently
HI CUT	Determines cut-off frequency of the high-cut filter applied to both channels
LO CUT	Determines cut-off frequency of the low-cut filter applied to both channels

Pitch Fix

Pitch Fix is a real-time pitch correction processor. Individual notes can be disabled by clicking on the corresponding piano keys—when a note is disabled, any input falling within that pitch range remains unaltered.

Figure 150: Pitch Fix

Parameter	Function
SPEED	Determines how fast the out-of-tune notes reach the target pitch
AMOUNT	Determines how much pitch-shifting is applied

Double Vocal

DOUBLE VOCAL is a simple and effective doubler effect with five operating modes. Although originally designed for vocals, it can also add a pleasing doubling effect to other instruments.

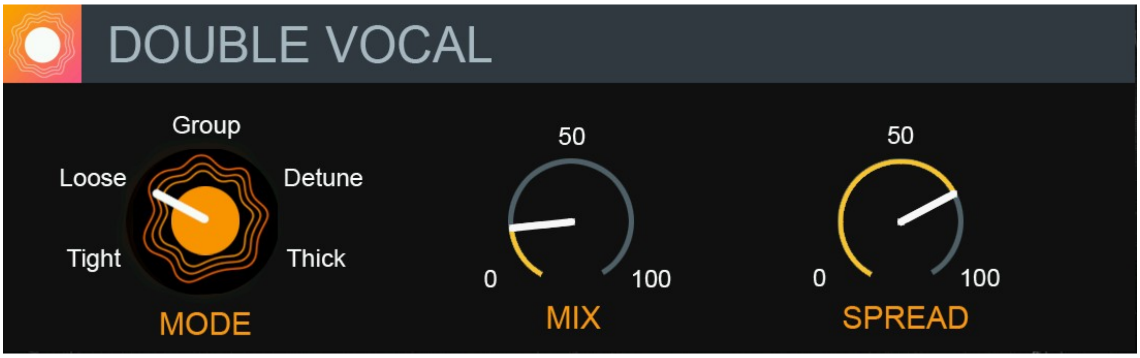

Figure 151: Double Vocal

Parameter	Function
MODE	Selects operating mode: Tight, Loose, Group, Detune, Thick
MIX	Adjusts balance between the processed (100%) and unprocessed (0%) signal
SPREAD	Adjusts stereo width of the processed signal

11.8 Aural Exciters

Exciter

Exciter is based on an analog aural exciter introduced in the mid-1970s, which added harmonic content to the signal while maintaining a constant output level.

Figure 152: Exciter

Parameter	Function
TUNE	Sets cut-off frequency of the low-cut filter applied to the signal fed into the harmonic generation stage (see Fig. 153)
PEAK	Emphasizes both the TUNE frequency and the preceding dip, functioning similarly to a resonance control in an EQ filter (see Fig. 154)
ZERO FILL	Reduces sharpness of the dip preceding the TUNE frequency, resulting in a smoother transition (see Fig. 155)

Parameter	Function
TIMBRE	Negative values add odd harmonics, while positive values add even harmonics
HARMONICS	Adjusts overall amplitude of the harmonics added. Note that the effect can still add brightness to the processed signal even when the HARMONICS knob is all the way down
DRY	When engaged, unprocessed signal is sent to the effect output
MIX	Adjusts balance between the processed (100%) and unprocessed (0%) signal

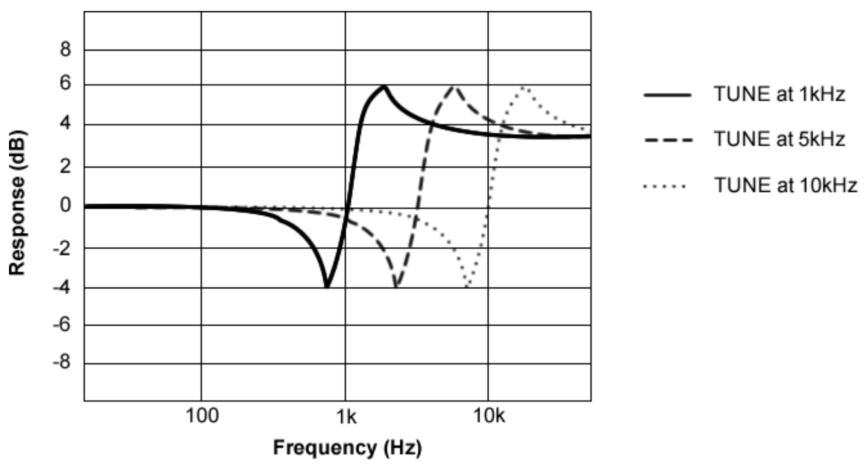

Figure 153: Frequency response for different TUNE values

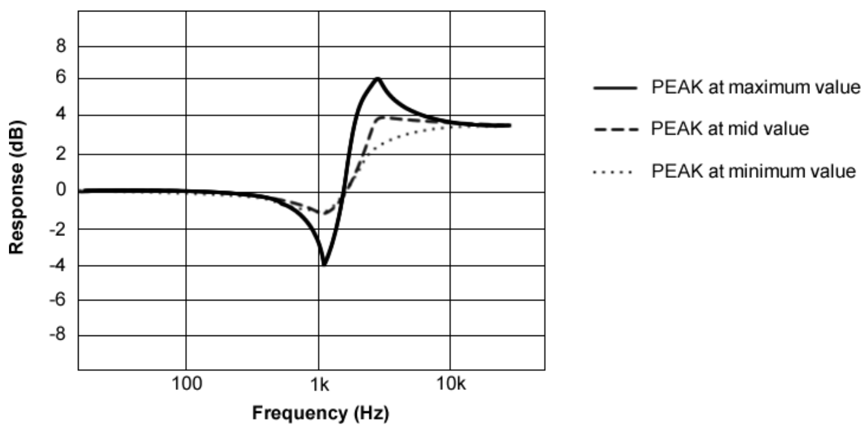

Figure 154: Frequency response for different PEAK values

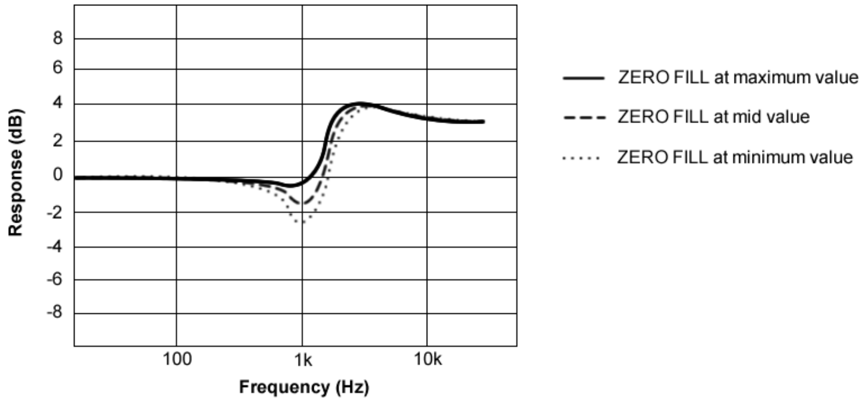

Figure 155: Frequency response for different ZERO FILL values

Ultra Enhancer

Ultra Enhancer is based on the tube version of a German stereo processor from the 1990s, designed to "unmask" sounds, resulting in more intelligible and detailed mixes.

Figure 156: Ultra Enhancer

Parameter	Function
FREQ (BASS)	The low frequency range is accentuated, dependent on this frequency value
GAIN (BASS)	Sets gain applied to the low frequency range
Q	Adjusts width of the mid band range
GAIN (MID)	Sets gain applied to the mid band range
FREQ (HIGH)	The high frequency range is accentuated, dependent on this frequency value
GAIN (HIGH)	Sets gain applied to the high frequency range
LMF SPREAD	Adjusts stereo width of the low-mid frequency content
LEVEL (STEREO)	Adjusts level of the stereo information
PAN (STEREO)	Adjusts panning of the stereo information
LEVEL (MONO)	Adjusts level of the mono content
PAN (MONO)	Adjusts panning of the mono panning

Parameter	Function
SOLO	Only additional "enhanced" content is sent to the effect output, excluding the unprocessed signal
GAIN	Adjusts overall output level

Tape Machine

Figure 157: Tape Machine

Parameter	Function
OUT GAIN	Adjusts effect's output level
LOW BUMP	When enabled, emulates the characteristic low-frequency irregularities that occur during playback on analog tape machines
HIGH SHELV	Enables processing of the high-frequency range, which is typically attenuated at lower tape speeds
DRIVE	Adjusts amount of drive added by the tape
SPEED	Adjusts the speed where the tape machine operates. Lower speeds result in a darker sound with less high frequency content

SOUL Warmth Pre

SOUL Warmth Pre emulates the preamps on a British console and offers many tonal options from subtle coloration to aggressive saturation. It is available in the GATE slot on all input channels.

Figure 158: SOUL Warmth Pre

Parameter	Function
HARMONICS	Controls blend of second order (even) and third order (odd) harmonics added
DRIVE	Adjusts amount of drive added
TRIM	Adjusts effect's output level
COLOR	Adjusts high-frequency roll-off that gradually attenuates the harmonics as they increase in frequency
MIX	Adjusts balance between the processed (100%) and unprocessed (0%) signal

11.9 Subharmonic Synthesizers

Psycho Bass

Psycho Bass is a low-frequency enhancer that adds harmonics to the signal, creating the illusion of bass frequencies that are not actually being played back.

Figure 159: Psycho Bass

Parameter	Function
INTENSITY	Adjusts level of the added harmonics
BASS GAIN	Attenuates signals below the X/O FREQUENCY to compensate for the perceived volume increase caused by the added harmonics
X/O FREQUENCY	Sets frequency, when below, the BASS GAIN is applied
SOLO	When enabled, only the frequency range below the X/O frequency is sent to the effect output

Sub Octaver

Sub Octaver is based on an early 1980s guitar pedal. It generates two new tones an octave and two octaves below the original sound.

Figure 160: Sub Octaver

Parameter	Function
OCTAVE 1	Adjusts level of the tone one octave below the original sound
OCTAVE 2	Adjusts level of the tone two octaves below the original sound
RANGE	Optimizes effect for different input sources. For example, LOW is better suited for bass or low notes on a guitar, while HIGH is ideal for the higher-ranged instruments

Sub Monster

The MIDAS SUB MONSTER plug-in is a subharmonic synthesizer that accentuates a specific frequency along with two harmonics above and two harmonics below it.

The center frequency is set using the Tune knob. Each fader controls the level of its corresponding harmonic. Note that the center frequency is found on Band 3.

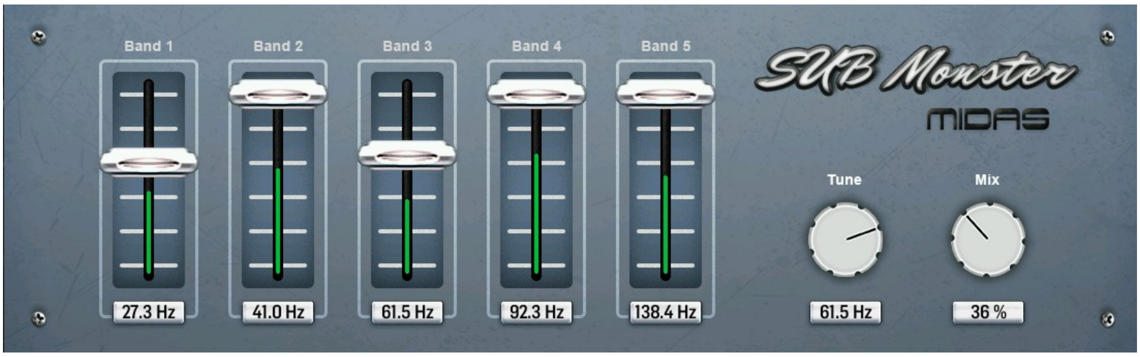

Figure 161: Sub Monster

11.10 Amp Simulations

Rack Amp

Rack Amp is inspired by an American-made analog processor that emulates guitar preamps, power amps and speakers.

Figure 162: Rack Amp

Parameter	Function
OUT GAIN	Controls overall output gain of the effect
DRIVE	Adjusts amount of power amp distortion
PRE AMP	Adjusts gain of the signal going from the preamp to the power amp
BUZZ	Controls low-frequency distortion
PUNCH	Controls mid-frequency distortion
CRUNCH	Controls high-frequency distortion
CABINET SIMULATION	Enables or disables speaker emulation
HI EQ	Adjusts gain applied by a high-shelving filter
LO EQ	Adjusts gain applied by a low-shelving filter

UK Rock Amp

UK Rock Amp is inspired by the tone of classic British guitar amplifiers commonly used for rock.

Figure 163: UK Rock Amp

Parameter	Function
CABINET	Enables or disables speaker emulation
OUTPUT GAIN	Controls overall output gain

Parameter	Function
SAG	Emulates the behavior of Class AB tube power amplifiers. At high volumes, incoming signals with hard transients (fast attack and high amplitude) cause a sudden increase in current draw. This causes the power supply voltage to drop slightly before recovering, creating a swell-like effect in the sound
GAIN	Adjusts preamp gain
BASS	Adjusts level of the low-frequency range
MIDDLE	Adjusts level of the mid-frequency range
TREBLE	Adjusts level of the high-frequency range
PRESENCE	Adjusts EQ boost applied by high-shelf filter at the power amp stage
MASTER	Adjusts power amp gain

ANGEL Amp

ANGEL Amp is based on a guitar amplifier commonly used for metal and other styles that benefit from an aggressive and distorted guitar tone.

Figure 164: ANGEL Amp

Parameter	Function
Output Gain	Controls overall output gain
SAG	Emulates the behavior of Class AB tube power amplifiers. At high volumes, incoming signals with hard transients (fast attack and high amplitude) cause a sudden increase in current draw. This causes the power supply voltage to drop slightly before recovering, creating a swell-like effect in the sound
Clean Gain	Adjusts preamp gain
Bass	Adjusts level of the low-frequency range
Middle	Adjusts level of the mid-frequency range
Treble	Adjusts level of the high-frequency range
Master	Adjusts power amp gain

Parameter	Function
Presence	Adjusts EQ boost applied by high-shelf filter at the power amp stage
MID Boost	When enabled, mid-frequency range is boosted
Bright	When enabled, high-frequency range is boosted
Bottom	When enabled, low-frequency range is boosted
Cabinet	Enables or disables speaker emulation

Jazz Clean Amp

JAZZ CLEAN AMP is inspired by a classic mid-1970s amplifier, famous for its clean tone and analog chorus effect. Despite its name, this solid-state amp is suitable for a wide range of genres.

Figure 165: Jazz Clean Amp

Parameter	Function
CABINET	Enables or disables speaker emulation
BRI	When enabled, high-frequency range is boosted
VOLUME	Adjusts preamp gain
TREBLE	Adjusts level of the high-frequency range
MIDDLE	Adjusts level of the mid-frequency range
BASS	Adjusts level of the low-frequency range
OUTPUT GAIN	Controls overall output gain

Deluxe Amp

Deluxe Amp is based on a famous 1960s American-made amplifier. Like the original tube amp, this effect can deliver guitar tones ranging from crystal clear cleans to crunchy drives.

Figure 166: Deluxe Amp

Parameter	Function
CABINET	Enables or disables speaker emulation
OUTPUT GAIN	Controls overall output gain
SAG	Emulates the behavior of Class AB tube power amplifiers. At high volumes, incoming signals with hard transients (fast attack and high amplitude) cause a sudden increase in current draw. This causes the power supply voltage to drop slightly before recovering, creating a swell-like effect in the sound
VOLUME	Adjusts preamp gain
TREBLE	Adjusts level of the high-frequency range
BASS	Adjusts level of the low-frequency range

BODYREZ

BODYREZ is based on the pedal by TC Electronic. It is designed to restore the natural acoustic resonance of acoustic guitars, especially when using sub-optimal pickups.

The more pristine sound is achieved by adding pre-defined filters and compression with a single knob.

Figure 167: BODYREZ

11.11 Miscellaneous

Speaker Manager

Speaker Manager is divided into three sections: FILTER, DYN EQ, and LIMITER. It is designed to apply essential processing to output signals for PA systems, stage monitor wedges, or in-ear monitoring (IEM) systems.

Figure 168: Speaker Manager

Speaker Manager FILTER page parameters

Parameter	Function
High pass	Sets cut-off frequency of the low-cut filter
Filter Type	Selects low-cut filter type
Tilt EQ	Adjusts tilt EQ gain
Tilt Freq	Selects tilt EQ cut-off frequency
Low Pass	Sets cut-off frequency of the high-cut filter
Filter Type	Selects high-cut filter type
Phase	Offsets the signal's phase by 0° to 180° degrees
Polarity	Inverts the signal's polarity
Precision Delay	Adjusts the delay applied to both channels
Position	Adjusts delay between left and right channels, corresponding to the distance between each speaker and the listening position. Negative values correspond to the left speaker being closer to the listener and vice-versa.

Speaker Manager DYN EQ page parameters

Parameter	Function
Attack	How quickly the processor applies the gain adjustment after the signal crosses past the threshold—either above or below, depending on the setting of the Above/Below switch
Release	How quickly the processor stops applying gain adjustment after the signal crosses back past the threshold—either below or above, depending on the setting of the Above/Below switch
Threshold	Level the signals must exceed for the equalizer to engage
Gain	Maximum gain adjustment applied by each band
Freq	Center frequency for parametric filters or cutoff frequency for shelf filters
Q	Width of frequency range affected by parametric filters only. This parameter has no effect on other filter types
Ratio	Reduces output level relative to how much the input exceeds the threshold. For example, at a 2:1 ratio, every 2 dB past the threshold results in only a ±1 dB change at output. The maximum gain adjustment is determined by the Gain parameter
Filter Types	Bands can be configured as a low or high Shelf filter, Parametric, or Flat. When Flat is selected, the band functions as a full-range dynamics processor. For shelving filters, 6 dB or 12 dB per octave selectable slope options of are available
Above / Below	Determines whether the unit applies the gain adjustment when the input signal exceeds the threshold (Above) or falls below the threshold (Below)

Speaker Manager LIMITER page parameters

Parameter	Function
Threshold	Level where the compressor will apply gain reduction
Release	Defines time taken to restore approximately two-thirds of the applied gain reduction
Active	Enables or bypasses the effect
RMS / Peak	Toggles between RMS and peak level detection

External

Use External to insert outboard digital or analog processors into the channel's signal flow.

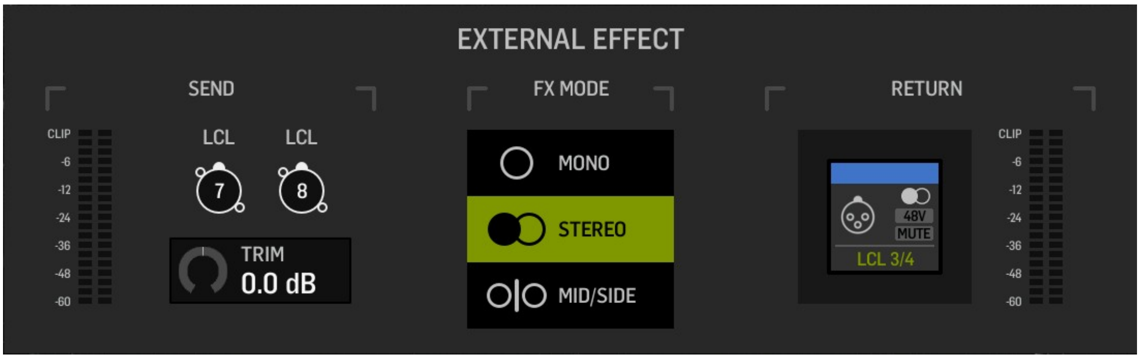

Figure 169: External

Parameter	Function
SEND	Clicking on this section opens the ROUTING › OUTPUTS screen. From there, assign the corresponding channels from the FX SENDS source group to the outputs connected to the outboard effect
TRIM	Adjusts send level
FX MODE	Selects from MONO, STEREO or MID/SIDE configuration
RETURN	Pressing opens a window to select the console inputs to receive the output of the outboard effect
LATENCY	Adjusts the delay in ms applied to the dry signal to keep it in phase with the wet signal

Section 12

Specifications

Processing	
Input processing channels	40 stereo input channels, 8 stereo aux channels
Output processing channels	16 stereo aux buses, 8 stereo matrices, 4 stereo mains
Internal effects engines (all true stereo)	8 premium FX slots, 8 standard FX slots
Point-to-point routing matrix	500 × 502 signals
Signal processing	40-bit floating point, 48 kHz
A/D converters (8-channel, 48 kHz, 24 bits)	114 dB dynamic range
D/A converters (stereo, 48 kHz, 24 bits)	120 dB dynamic range
I/O latency (console input to output)	1.0 ms
Network latency (Stagebox input → Console → Stagebox output)	1.2 ms
Connectors	
Midas PRO series microphone preamplifier (XLR/TRS combo jack)	8 (WING), 24 (COMPACT and RACK)
Midas PRO series XLR balanced outputs	8
Aux inputs/outputs (1/4" TRS balanced, mono)	8 in / 6 out + 2 monitor or headphone outputs (WING), N/A (COMPACT and RACK)
Phones output (1/4" TRS, stereo)	2 (WING and COMPACT), 5 (RACK)
Digital AES/EBU input/output (XLR)	1 / 1
AES50 ports (Klark Teknik SuperMAC, 100 Mbit/s)	3
Expansion card interface	64 × 64 channel audio input / output
StageConnect HOST(Master) I/O (12 V / 18 W power supplied, XLR, 32 channels)	1
MIDI inputs/outputs	1/1
GPIO on TRS, configurable	2×2 (WING and RACK), 1×2 (COMPACT)
USB 2.0 type B device (48 × 48 ch 24-bit audio and MIDI I/O)	1
USB 2.0 type A host (audio and data, 5 VDC, 1 A)	1

Ethernet LAN ports, RJ45, 1 Gbit/s	2, internally switched
Audio over IP (AoIP) internal module socket (Dante or SoundGrid modules optional)	Up to 64 × 64 channels @ 48 kHz
IEC mains socket with power switch	1
Mic Input Characteristics (mic input to analog output)	
Design	Midas PRO series
THD+N (0 dB gain, 0 dBu output)	<0.004%
THD+N (+40 dB gain, 0 dBu to +20 dBu output)	<0.006%
Input impedance (unbalanced / balanced)	1 $k\Omega$ / 2 $k\Omega$
Non-clip maximum input level	+21 dBu
Phantom power (switchable per input)	+48 V
Equivalent input noise @ +45 dB gain (150 Ω source)	-128 dBu
CMRR @ unity gain (typical)	>50 dB
CMRR @ 40 dB gain (typical)	>70 dB
Input / Output Characteristics	
Frequency range	10 Hz - 20 kHz (± 1dB)
Dynamic range, analog in to analog out (typical), XLR / AUX	111 dB / 108 dB
A/D dynamic range, preamplifier and converter (typical), XLR / AUX	112 dB / 110 dB
D/A dynamic range, converter and output (typical), XLR / AUX	118 dB / 112 dB
Crosstalk rejection @ 1 kHz, adjacent channels	100 dB
Output level, XLR connectors (nominal / maximum)	+4 dBu / +21 dBu
Output impedance, XLR connectors (unbalanced / balanced)	75 Ω / 75 Ω
Input impedance , TRS connectors (unbalanced / balanced)	20 $k\Omega$ / 40 $k\Omega$
Non-clip maximum input level, TRS connectors	+16 dBu
Aux output level, TRS (nominal / maximum)	+4 dBu / +16 dBu
Aux output impedance, TRS (unbalanced / balanced)	150 Ω / 300 Ω
Headphone output impedance / maximum output level	500 mW @ 75 Ω / +18 dBu
Residual noise level, XLR out 1-16 connectors, unity gain	-97 dBu
Residual noise level, aux and monitor TRS out connectors	-95 dBu
Displays	
Main screen	10.1" TFT LCD, 1280 × 800 px, capacitive touch
Main screen swivel, continuous adjustment	15°- 60° (WING and COMPACT), 0°- 45° (RACK)

4-channel group LCD screen with RGB color strip per channel	320 x 48 monochrome (WING and RACK), N/A (COMPACT)
Channel editing screen	2.4" TFT LCD, 320 × 240 px (WING), N/A (COMPACT and RACK)
Button assignment screen	2.4" TFT LCD, 320 × 240 px (COMPACT), N/A (WING and RACK)
Main stereo meter	18 segment (-60 dB to clip)
Controls	
100 mm motor faders	12 + 8 + 4 (WING), 12 + 1 (COMPACT), N/A (RACK)
Touch-sensitive rotary controls	3 + 7 + 11 + 4 + 4 (WING), 7 + 3 (COMPACT), 1 + 4 + 1 (RACK)
Fully assignable rotary CUSTOM CONTROLS	4 (WING and RACK), N/A (COMPACT)
Fully assignable backlit button CUSTOM CONTROLS	8 + 8 (WING), 16 (COMPACT), 8 (RACK)
Variable rotary / backlit button CUSTOM CONTROLS	4 / 4 (WING), N/A (COMPACT and RACK)
Power	
Switch-mode power supply	Auto-ranging 100-240 VAC (50/60 Hz)
Power consumption	130 W
Physical	
Standard operating temperature range	5°C – 45°C (41°F – 113°F)
Dimensions (H × W × D)	WING 201×870×575mm (7.9×34.3×22.6") COMPACT 225×453×574 mm (8.9×17.8×22.6") RACK 183×326×486 mm (7.2×12.8×19.1")
Weight	WING 24 kg (52.8 lbs) COMPACT 15.4 kg (34 lbs) RACK 15.4 kg (34 lbs)

Note: all noise and dynamic range measurements are A-weighted.