m908 multi channel monitor controller

user manual

Revision P / 03/29/2023

ACU Firmware version: 2.1.0

RCU Firmware version 2.0.0

Please visit https://gracedesign.com/support-documents/ for the latest m908 firmware and documentation. See the Firmware Update Procedure section of this manual for details.

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Table of Contents

List of Tables

1   Welcome

Thanks for purchasing the m908 multi-channel monitor controller. It has been carefully designed and built to provide you with a beautiful sounding, configurable and reliable monitoring system. Please familiarize yourself with the setup and operational details contained in this manual. And as always with this or any other Grace Design products, please don't hesitate to reach out if you have any questions. We are available by telephone Monday – Friday, 9AM to 5PM MST, or by email at info@gracedesign.com. Also, other information including technical documents and firmware can always be found on our website – www.gracedesign.com. Thanks for reading and enjoy your m908!

2   Important Safety Information

2.1 General

- Indoor use only
- Ordinary Protection: This equipment should not be exposed to dripping or splashing.
- Avoid placing objects filled with liquids, such as vases or glasses, on this equipment.
- Class I Equipment (grounded type)
- Electrical rating: 90-240V~ 50-60Hz 50W
- Mains supply voltage fluctuations are not to exceed ±10% of the nominal supply voltage.
- Pollution Degree 2
- Installation (Over voltage) Category II for transient overvoltages.
- Maximum Relative Humidity: <80%
- Operation temperature range: 10 °C to 40 °C
- Storage and transportation temperature range –40 °C to 70 °C
- Maximum altitude: 3000m (9843 ft)
- Equipment suitable for continuous operation

2.2 Safety Marking Symbols

This symbol, located on the equipment and in this manual, refers to important instructions. Read this manual thoroughly before operating this equipment.

This symbol, located on the equipment and in this manual, indicates the potential for electrical shock hazard.

2.3 Service Information

The Grace Design m908 contains no user serviceable components. Contact Grace Design for repair and upgrade information. In the event that your Grace Design m908 needs to be returned to the factory, contact us for a return authorization number.

2.4 California Proposition 65 Warning

This product may contain metallic nickel, a chemical known to the State of California to cause cancer and birth defects or other reproductive harm. Wash hands after handling.

2.5 Support

The m908 is a powerful and complex monitoring system. If you have any problems configuring or operating it please do not hesitate to email or call our customer service department.

service@gracedesign.com
303-82308100 x105

The current version of this manual, related documents, and the latest m908 firmware can be found at:

https://gracedesign.com/support-documents/

3   Overview And Features

3.1 Features

- 24 Channel DSP for immersive surround formats such as ATMOS™, DTS:X™ and Auro 3D™
- Manage playback systems from mono through 22.2
- 16 User definable input sources
- Summing of up to 3 input sources
- 3 user definable Control Room speaker outputs
- Our latest generation of AD and DA converters
- Powerful and ergonomic remote control unit for access to all system controls
- 4th generation s-Lock pll clocking system for vanishingly low jitter
- High resolution volume control
- Up to 1 second sync delay
- 12 bands of Room correction EQ on each channel
- Full bass management
- Speaker channel level and delay calibration
- Comprehensive downmix control
- 16 channel analog out / 24 channel AES3 digital I/O

- 16 channel ADAT Lightpipe In / USB 24 channel In
- AES3, S/PDIF, and TOSLINK Stereo In
- Optional Dante™, Ravenna/AES67, or DigiLink™ modules for additional 32 channel I/O
- Optional 8 channel ADC module for 8 or 16 channel analog inputs
- Optional high performance RIAA phono preamplifier for MC and MM cartridges.
- Flexible Meter outputs
- Built in SPL meter
- Dual redundant external power supply
- Reference quality headphone amplifier with cross-feed.
- Flexible talkback system with built-in mic on remote control and mic input for external talkback mics
- Up to 8 stereo CUE paths
- 4 General Purpose Input and Output pins for interfacing to external systems.
- Intuitive and powerful web UI for configuration and control.

The m908 consists of three components: ACU (Audio Control Unit), RCU (Remote Control Unit), and PSU (Power Supply Unit) Below is a description of the connections, controls, and indicators for each component.

3.2 ACU Description

The 2U ACU chassis houses all of the analog and digital audio IO, DSP processing, and control IO. The audio signal processor is a 1GHz 32 bit fixed point unit which allows for highly accurate and low latency signal processing. 2GB of DDR2 ram allows for ample sync delay across 24 channels at sample rates up to 192kHz. Audio routing, AES and ADAT encoding and decoding is handled by an Artix-7 FPGA. The system is managed by an ARM Cortex-A15 processor running embedded Linux.

The m908 system clock is based on our 4th generation s-Lock PLL technology. This is an incredibly low jitter PLL that provides exceptional ADC and DAC sample clocking accuracy.

The Audio and Clock flow is shown in the simplified block diagram below.

GRACE
DESIGN
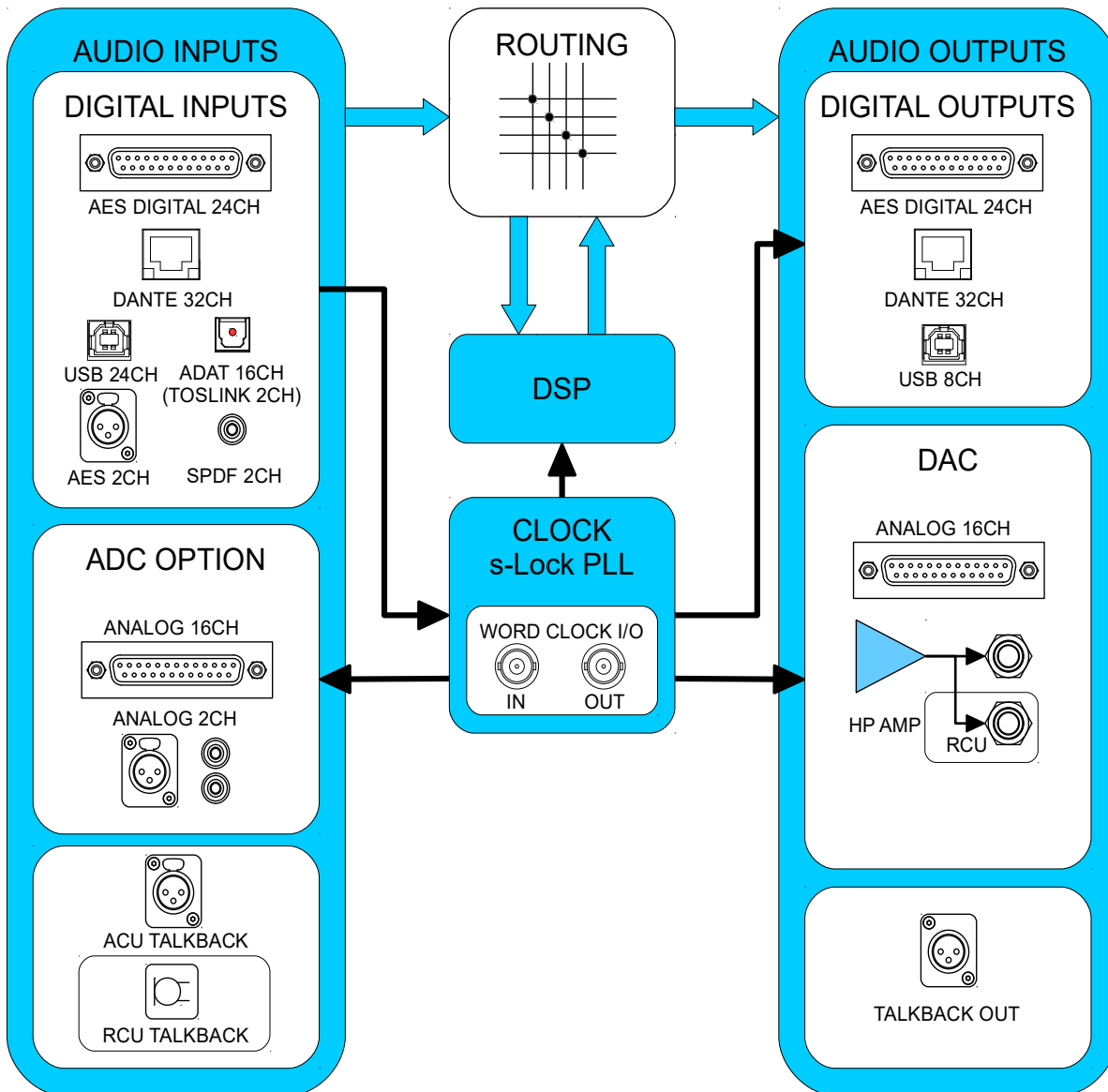
Illustration 1: Audio and Clock Flow Diagram

The audio signal processing signal flow is shown in the diagram below.
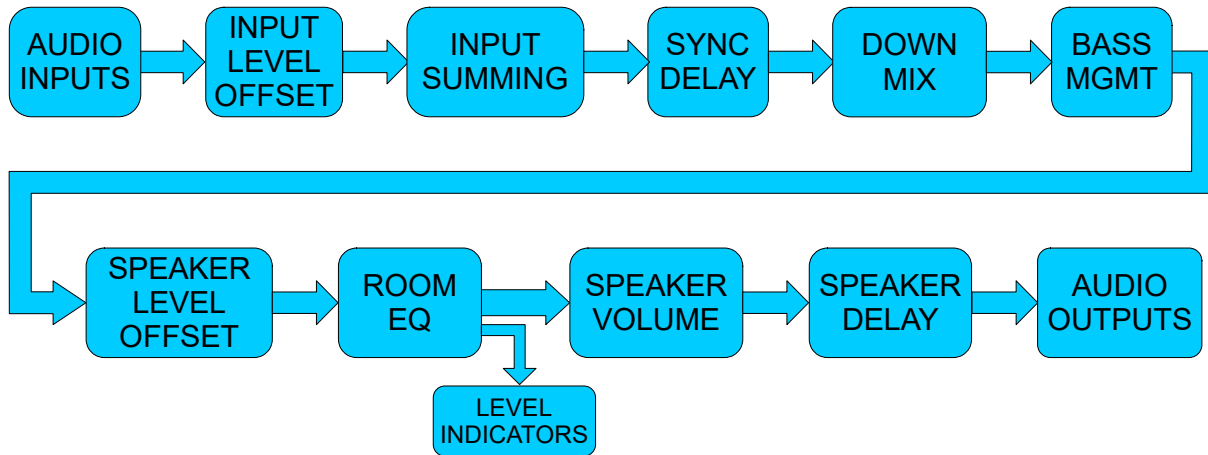
Illustration 2: Audio Processing Flow Diagram

3.3 ACU Front Panel Connections and Controls
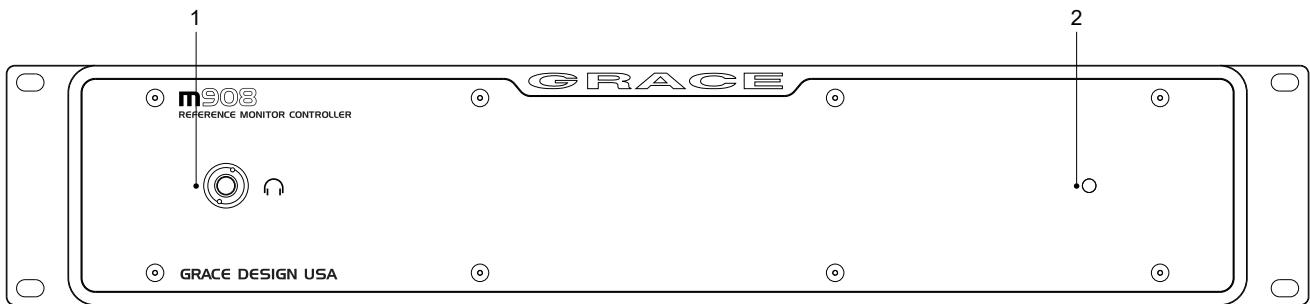
Illustration 3: ACU Front Panel

1. 1/4" Headphone Jack: The m908 headphone amplifier is an ultra high performance current feedback design. Very low output impedance ensures good damping factor at the headphone driver. The crossfeed circuit provides natural imaging and minimizes listening fatigue.
2. Power indicator LED: Green LED illuminates solid green under normal conditions.

3.4 ACU Rear Panel Connections
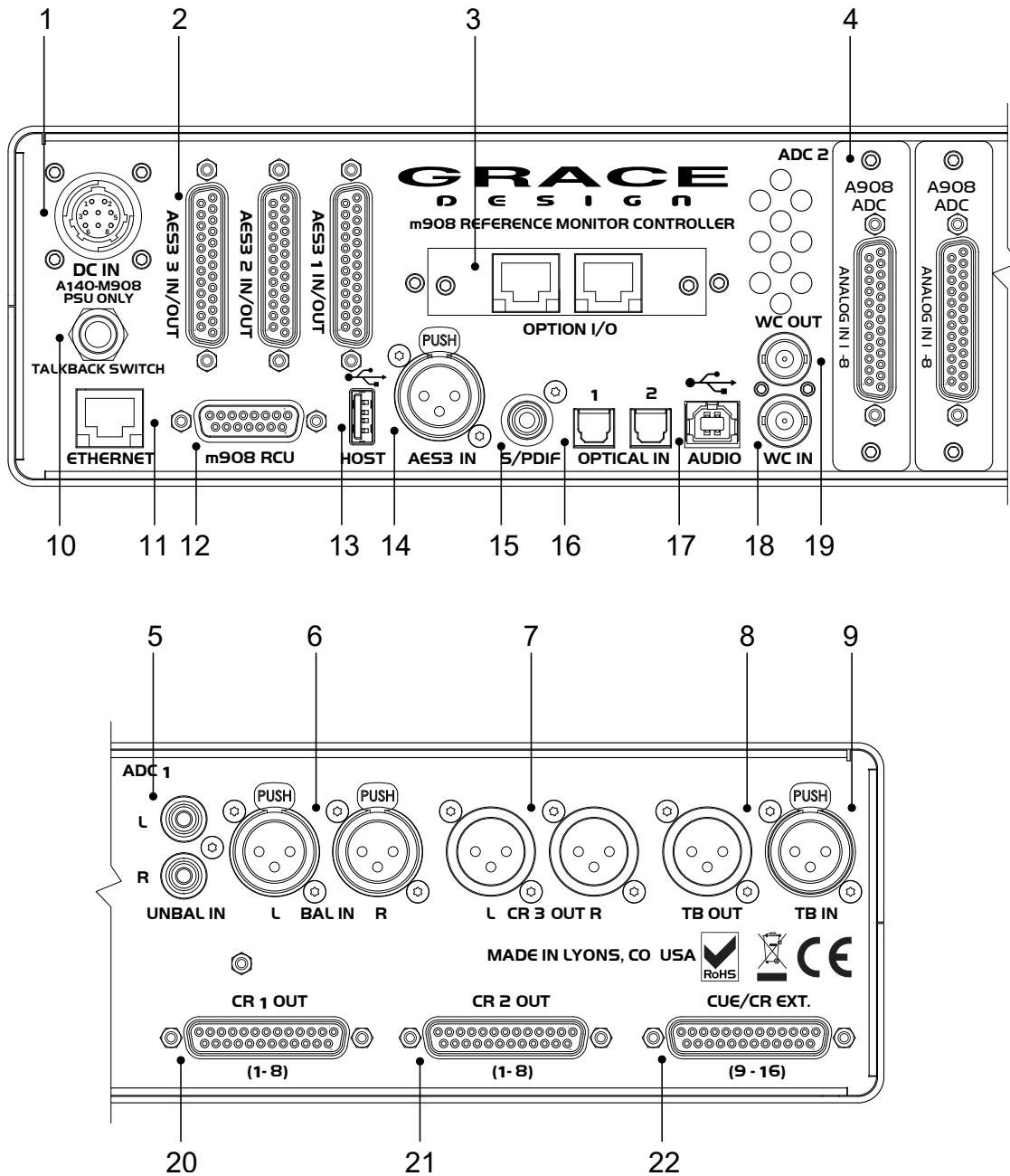
Illustration 4: ACU Rear Panel

1. DC power input connector 8 Pin Circular: Connect to PSU. CAUTION: CONNECT PSU TO ACU WITH DC POWER CABLE BEFORE APPLYING POWER.

2. AES3 I/O x 3 DB25 (Tascam pinout) 110Ω: 24 channel input and output. Accepts sample rates from 44.1kHz-192kHz.Each DB25 includes a single GPIO signal.

3. Option I/O slot (Dante , Digilink, and Ravenna-AES67 options): The Option I/O module can receive and transmit 32 channels of PCM audio.

4. Analog balanced input ADC (optional): There are slots for two optional 8 channel ADC modules with DB25 input connectors. The ADC modules plug in to the main digital pcb. Two ADC modules can be used for a total of 16 channels. The ADC in position 1 is also connected to the stereo balanced XLR and unbalanced phono jacks so that it can be used for monitoring stereo sources on channels 1-2. Note that when an input uses the stereo connections, channels 1-2 on the DB25 connector are not available for assignment.

5. Stereo Input analog unbalanced RCA Phono: Present if an ADC option module is installed. Can be switched to Phono level if the RIAA Phonograph Preamplifier module is installed.

6. Stereo Input balanced analog XLR: Present if an ADC option module is installed.

7. Control Room 3 Output balanced analog XLR: Analog output signals sourced from DAC channels 1-2.

8. Talkback output balanced analog XLR: Talback signal sourced from ACU or RCU Mic. Can be configured to be momentary while talkback is active or continuously on.

9. Talkback mic input balanced XLR: Balanced microphone level input with available 48V phantom power. The talkback signal is fed to an ADC and sent to the DSP section and can also sent to the Talkback output XLR connector when talkback is active.

10. External Talkback switch jack 1/4" TRS: Wiring a normally open switch to this jack will allow for another talkback control. The Ring of this connector carries current limited 5V for powering a tally LED. See Cable Diagrams.

11. Ethernet Control Port RJ45: -Network connection for web UI. See Web-Network Setup.

12. RCU serial control DB15: Connect to RCU with supplied DB15 cable.

13. USB Host USB-A: USB host connector for system firmware update.

14. AES3 Input XLR: Stereo AES digital input. Supports sample rates from 44.1kHz-192kHz.

15. SPDIF Input RCA Phono

16. ADAT/TOSLINK x 2 Optical: Optical receivers for ADAT or TOSLINK. Supports SMUX for 8ch 96kHz operation.

17. USB Audio I/O USB-B: The USB Audio Class 2 compliant interface allows for monitoring up to 24 channels of PCM audio at 48kHz/24bit and 8 channels of PCM audio at 192kHz/24bit.

18. Word Clock IN BNC 75Ω or 47kΩ (setup menu selected): External clock input.

19. Word Clock OUT / THROUGH BNC 75Ω: Output clock signal can be a buffered copy of the incoming word clock or can be from the internal word clock.

20. Control Room 1 output balanced analog DB25 (Tascam pinout): Analog output signals sourced from DAC channels 1-8.

21. Control Room 2 output balanced analog DB25 (Tascam pinout): Analog output signals sourced from DAC channels 1-8

22. CUE/WIDE OUT balanced analog DB25 (Tascam pinout): Analog output signals sourced from DAC channels 9-16. These outputs can be for speakers in immersive playback systems or for CUE outputs. CUE outputs have level control, assignable talkback summing, and MON>CUE routing. MON>CUE sends the Left and Right signal from the current monitor signal to the CUE output. In the case of multi-channel monitoring the Left and Right signal or a stereo downmix can be sent to the CUE output.

3.5 RCU Description

The Remote Control Unit is designed to be a seamless interface between the user and the audio system being controlled. We have carefully selected the display, switches, and encoder to be long life and provide excellent tactile feedback. All of the necessary information is displayed on the 1/4WVGA display while the most used functions have dedicated switches. There is a 1/4" headphone jack on the rear of the RCU for convenient access to the m908s high performance headphone amplifier.

The RCU sits on a tilting base so it can be adjusted to the most comfortable angle. The built in talkback microphone on the RCU also doubles as an accurate sound pressure level (SPL) meter.

The RCU is powered from the ACU via the DB15 cable.

3.6 RCU Front Panel Controls

Many of the most used monitoring functions are directly accessed through the push button switches on the RCU front panel. Primary functions are accessed with a press-release. Where a switch has more than one function the primary function is indicated with large upper case letters. The secondary function is indicated in small letters and is accessed with a push-hold of the switch. Some switches can have secondary functions assigned to them by the user. These switches do not have lower case labeling.
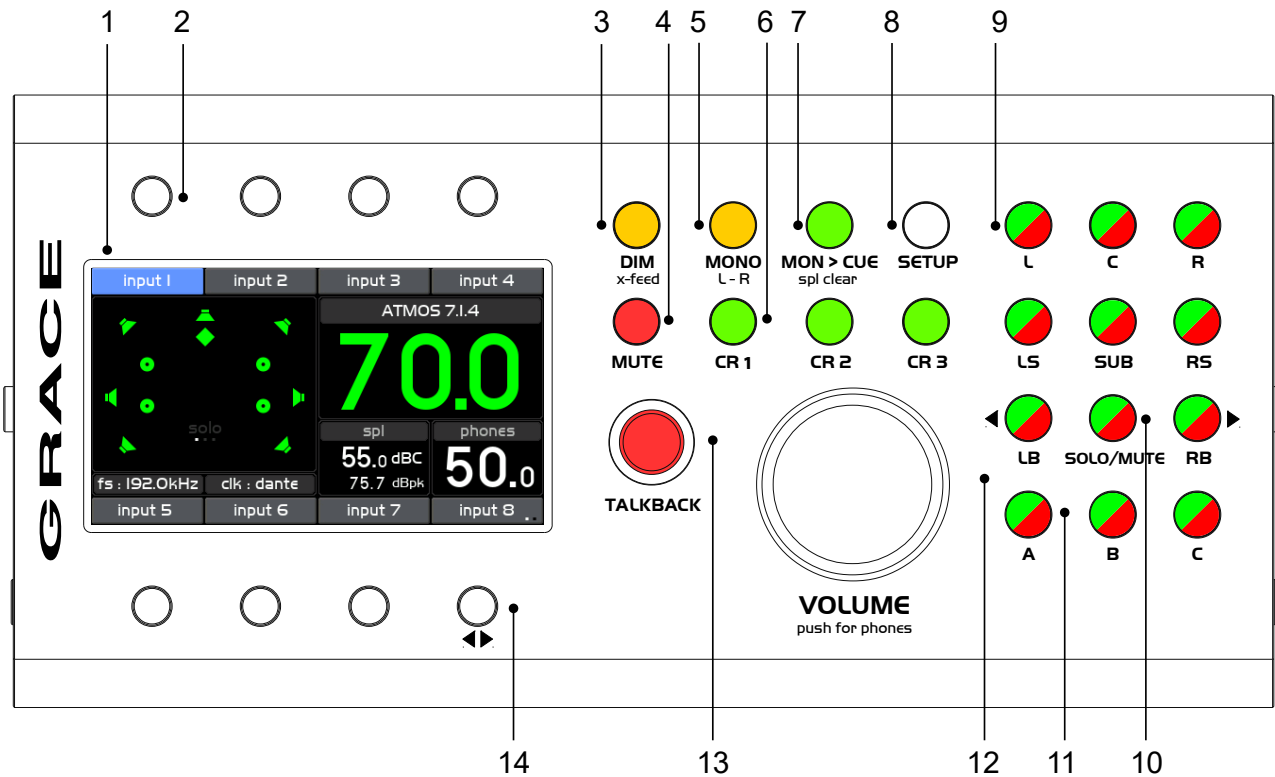
Illustration 5: RCU Front Panel

1. LCD Display screen: This color LCD displays all operating parameters and setup menus.
2. INPUT Select switches (8x): These switches select the input source to be monitored. See Input Select operation for details.
3. DIM (x-feed): Dims the monitor system by a pre-programmed level. Push-hold enables/disables the headphone cross-feed circuit. See DIM operation for details.
4. MUTE: Mutes the monitor system. See MUTE operation for details.
5. MONO (L-R): Sums the Left and Right channels (L+R). Push-hold for difference (L-R). See MONO operation for details.
6. CR1, CR2, CR3: Control Room Speaker system select. See CR1 CR2 CR3 operation for details.
7. MON > CUE (spl clear): Sends the current source being monitored to the CUE sends. Push-hold clears the maximum spl field in the sound pressure level meter. See MON>CUE operation for details.
8. SETUP: System Setup access. Push-Hold for System Workflow setup. See System Setup and Workflow Setup for details.

9. L, C, R, LS, SUB, LR, LB, RB: Speaker select for SOLO or MUTE. See SOLO MUTE for details.

10. SOLO/MUTE: Selects between SOLO and MUTE monitoring mode. See SOLO MUTE for details.

11. A, B, C: User programmable switches. See A B C User Switches operation for details.

12. VOLUME: Monitor level control for Control Room speakers and headphones. See VOLUME operation for details.

13. TALKBACK: Talkback is activated by pushing this switch. See TALKBACK operation for details.

14. INPUT Select switch and Input Page change. See Input Select operation for details.

3.7 RCU Rear Panel Connections
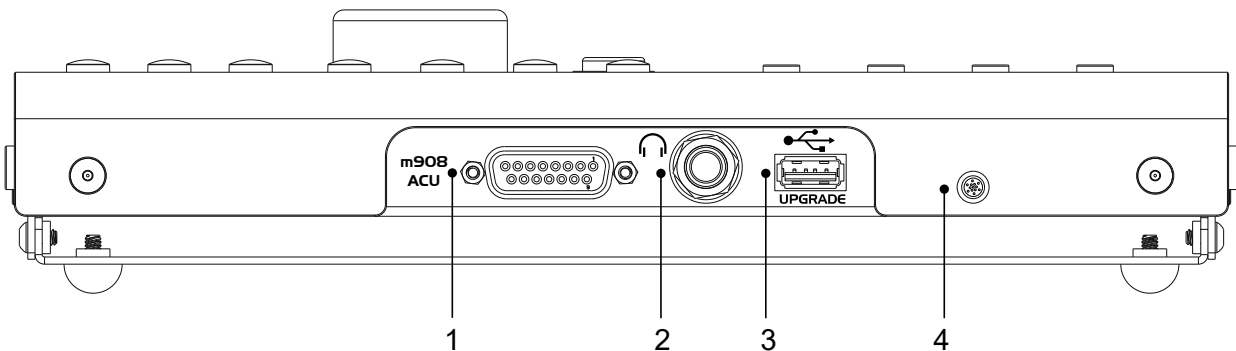
Illustration 6: RCU Rear Panel

1. m908 ACU DB15: Serial control cable connection to ACU
2. Headphone Jack 1/4" TRS: Carries stereo headphone signal from ACU headphone amplifiers. This jack is wired in parallel with ACU front panel headphone jack.
3. USB Host USB-A: USB host connection for user workflow export/import.
4. Microphone: Built in mic for talkback and spl measurement

3.8 RCU Tilting Base

The RCU features a tilting base so that it can be tilted to an angle for optimal viewing. Loosen the four thumbscrews located on each side of the RCU and tilt the unit to the desired angle. While holding the RCU at this angle tighten the four thumbscrews. The tilt range is from

horizontal to 30º.

3.9 PSU Description

The m908 monitor controller is powered by the A140 Dual Redundant Power Supply Unit. The A140 features Universal AC Input for operation from 100-240VAC. This new generation external power supply contains two complete power supply modules which operate in parallel. Should one power supply module fail the other module will continue to power your system with no interruption. The A140 contains an intelligent microprocessor that monitors all of the DC rail voltages and reports fault conditions by flashing front panel LEDs. In the unlikely event that a power supply module fails the A140 front panel LEDs with flash to indicate which power supply module is at fault. The power supply modules are easily replaced in the field which minimizes any possible system down time.

A low speed serial communication link allows the power supply unit to communicate voltage rail status to the main ACU processor.

Under normal operating conditions the four front panel Status LEDs will be solid green. If a DC output of power supply module #1 becomes too low or too high the corresponding LED will flash once repeatedly. If a DC output of power supply module #2 becomes too low or too high then the corresponding LED will flash twice repeatedly. If this happens contact Grace Design Technical Support at 303-823-8100 x105 or service@gracedesign.com

3.10 PSU Front Panel Controls and Indicators

Illustration 7: PSU Front Panel

1. Voltage rail monitor LED x 4: These LEDs will illuminate steady green under normal operation.
2. Mains power switch: This is the AC Mains disconnect device. It must remain accessible.

3.11 PSU Rear Panel Connections
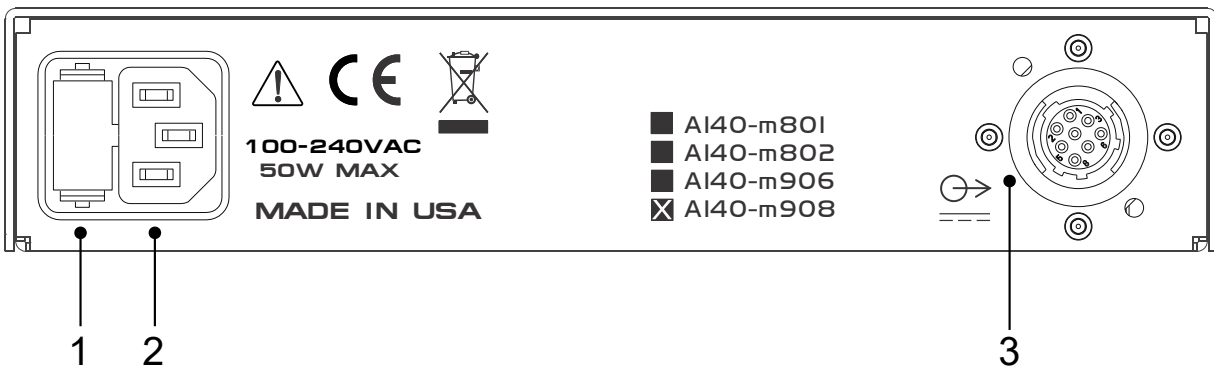
Illustration 8: PSU Rear Panel

1. Fuse drawer: Contains (2) 5x20mm 800mA Time Delay fuse.
2. AC Power inlet: 90-240V~ 50W max. Connect to a grounded outlet.
3. DC Output connector: ALWAYS connect DC power cable to m908 before turning on power and turn off the power before disconnecting the DC power cable!

4   Installation

4.1 Unboxing your m908

4.1.1   Open and inspect your box

Please take care when removing the m908 from its OEM packaging. Our packaging is designed with 'Korrvu©' suspension inserts, which provide the best protection from the dubious underworld of freight handling – so save your packaging material. Inner boxes will contain all the items listed below. If you think you are missing anything, give your dealer or us a call and we'll get you taken care of right away.

4.1.2   Your box will contain

- 1pc Audio Control Unit (ACU)
- 1pc Remote Control Unit (RCU)
- 1pc Power Supply Unit (PSU)
- 1pc AC power cord
- 1pc DC power cord 8' (2.8m) (Grace Part No. WA084)
- 1pc DB15 Remote Cable, 25' (7.6m) (Grace Part No. WA141)

- 1pc USB 2.0 Cable, Type A to Type B, 6 foot (Grace Part No. WA551)
- 2pc spare fuses. 800mA 250V~ Time Delay 5x20mm (Grace Part No. F101, Littlefuse 0239.800HXP or equivalent)
- 4pc Adhesive backed rubber feet, 0.12" thick (Grace Part No. H560)
- 4pc Adhesive backed rubber feet, 0.23" thick (Grace Part No. H580)
- 4pc 10-32 X 3/4 Truss Head Phillips rack mount screw (Grace Part No. H455)
- 1pc USB flash drive with m908 User Manual

4.1.3   REGISTER YOUR UNIT

We strongly urge you to fill out your warranty registration card. The m908 is covered by a 5 year transferable warranty. Registering your unit will help us contact you if there are important updates and simplifies warranty service. So please take a few minutes to complete and send in the enclosed card, or simply fill out the warranty registration form on our website. We always keep your information private. Thank you!

4.2 Connecting the m908

4.2.1   Power Connections

A DC power cord is supplied to connect the PSU to the ACU. This cord can be identified by the 8 pin circular connectors at each end.

Please note that the DC power cord should be connected _before_ the AC power is turned on. This prevents incorrect power sequencing which can cause damage to the audio circuits. To avoid any interference with the low level audio circuitry, the power supply should be located at least 3' (1m) from the ACU.

WARNING: A damaged DC power cord can create a shock hazard as voltages of 72VDC can be present.

Do not operate the m908 with a damaged DC power cord. If damage occurs, please contact Grace Design for a replacement. A standard AC power cable is included. For safety, the power supply cord must be connected to a grounded outlet.

4.2.2   Audio Connections

Talkback Microphone Input: Female XLR, pin 2 positive, pin 3 negative and pin 1 ground. 48V phantom power is supplied on pins 2 and 3. With a gain range +11dB and +19-74dB in 1dB steps this input can be used for any type of microphone.

Talkback Microphone Output: Male XLR, balanced, pin 2 positive, pin 3 negative and pin 1 ground. Provides a direct output of the talkback signal when talkback is active. See the Talkback setup section of this manual for details.

Analog Balanced Control Room Outputs: DB25 Tascam pinout (See CABLE DIAGRAMS). These outputs are defined in the setup menus for your Control Room speaker system.

Analog Balanced CUE/WIDE Outputs: DB25 Tascam pinout (See CABLE DIAGRAMS). Depending on your workflow these outputs are defined in the setup menus for your additional surround/overhead speakers or CUE outputs.

Analog Balanced Stereo Control Room Output: Male XLR, balanced, pin 2 positive, pin 3 negative and pin 1 ground. These outputs are defined in the setup menus for your Control Room speaker system.

AES3 Digital I/O: DB25 Tascam pinout, 110Ω (See CABLE DIAGRAMS)

ADAT Lightpipe / TOSLINK Inputs: In ADAT Lightpipe Mode these optical jacks provide 8 channels of audio data on each (at 44.1kHz and 48kHz). With ADAT S-MUX enabled 88.2kHz or 96kHz sample rate audio channels 1-4 are received on OPTICAL input 1, while channels 5-8 are received on OPTICAL input 2. 176.4kHz and 192kHz sample rates are not supported over the ADAT interface. In TOSLINK mode stereo signals up to 96kHz can be received on each connector. See the System Setup section of this manual for details on configuring the optical input connectors.

SPDIF Digital Stereo Input: RCA Phono jack, 75Ω. This input will receive PCM audio signal sample rates from 44.1kHz to 192kHz.

USB Class 2 Audio Interface: USB Type B jack. This interface is capable of 24 channel input from host computer and 8 channel output to the host computer. Use a standard USB type A -to- type B cable (included with your m908). The type A connector is to be plugged in to the HOST computer and the type B connector to the m908 input. PCM 44.1kHz – 192kHz/24 bit is supported. Please refer to USB Audio Setup.

4.2.3   Optional ADC Module Connections

Analog Balanced 8 Channel Line Inputs: DB25 Tascam Pinout (see CABLE DIAGRAMS). 8 channel ADC module balanced inputs. See ADC option details in the ADC Option Module section of this manual.

Analog Balanced Stereo Line Input: Female XLR, balanced, pin 2 positive, pin 3 negative and pin 1 ground. Supplied with ADC Option Module. Allows for the connection of balanced stereo sources to the ADC. Left channel is connected to ADC 1 channel 1 and Right channel is connected to ADC 1 channel 2.

Analog Unbalanced Stereo Line Input: RCA Phono. Supplied with ADC Option Module. Allows for the connection of unbalanced stereo sources to the ADC. Left channel is connected to ADC 1 channel 1 and Right channel is connected to ADC 1 channel 2.

4.2.4   Clock and Control Connections

REMOTE CONNECTOR: DB15 The m908 RCU handles all system control. Connection to the m908 RCU is via this DB15 connector, which carries RS485 serial data, DC power and headphone signals. The m908 ships with a high quality 25' cable. Custom cables can be assembled by the user according to the diagrams in the Cable Diagram section of this manual. While the serial data can travel over 1000 feet we do not recommend cables longer than 50' for headphone use. If you need a longer, cable contact your Grace Design dealer or us directly. Do NOT use an off the shelf DB15 cable as the pinout will be incompatible and may cause damage to your system.

WORD CLOCK IN: BNC 75Ω or 47kΩ (setup menu selected). The m908 can accept a standard 5V/75Ω word clock signal from an external clock generating unit. This might be a stand-alone clock source or via, for example, the word clock output from your digital audio workstation. The m908 locks to the incoming word clock with an ultra low jitter PLL. The PLL has a fast lock mode which rapidly acquires lock and then switches to a high jitter rejection mode with a loop bandwidth of 0.5Hz. This provides exceptional jitter rejection for the DAC and ADC sample clocks. In the event of a dropout or loss of incoming word clock signal, the intelligent PLL will remain at the last known valid frequency. When the signal is restored the PLL will smoothly re-acquire lock.

WORD CLOCK OUT: BNC 75Ω. The word clock output allows the user to synchronize other digital audio equipment to the m908. When the system menu setting "clock output" is set to external, this connector output is a buffered copy of the signal on the word clock IN jack. This mode is useful for "daisy chaining" to other units in a system. When "clock output" is set to system clock this connector outputs a buffered copy of the internal m908 word clock. The output is buffered and is designed to drive a 75Ω line.

ETHERNET: RJ45. This is a standard 1000 Base-T Ethernet interface. Connect this to a computer network to access the web UI. See Web-Network Setup for configuration details.

USB HOST: USB A connector. This host connector is intended for firmware upgrades.

EXT. TALKBACK: 1/4" TRS Jack. External talkback control. This input allows the connection of an external switch, such as a footswitch, for remotely activating the ACU talkback mic input, or the built in talkback mic on the RCU. The input is a TRS jack and is used with a "normally open" switching device. Note that when using this jack the associated talkback function can be assigned in talkback setup. See the remote talkback cable diagram at the end of this manual for connection details.

4.2.5   Thermal Management

The m908 ACU requires adequate ventilation to maintain proper operating temperature.

Under normal operating conditions the m908 will dissipate 30-35W of power depending on user settings and hardware options. To protect the m908 from excessively high internal

temperatures, and to ensure long term reliability, it is equipped with a 40mm variable speed fan. Whether or not this fan turns on is dependent on several variables. These are: ambient temperature, whether there is air movement around the m908, and if the m908 is mounted directly above or below other heat generating equipment.

The variable speed fan comes on to cool the internal circuits if the main circuit board temperature rises above 55ºC (131ºF). The fan has three speed ranges which are triggered by the following pcb temperature thresholds:

1. Low: 55ºC (131ºF)
2. Med: 60ºC (140ºF)
3. High: 65ºC (149ºF)

If the m908 is mounted without space above and below and it is adjacent to heat generating devices then the internal temperature can rise above 55ºC and the fan will come on.

If the m908 is being used in an area where fan noise is undesirable it is recommended that the m908 be mounted such that it has space below and above the chassis and where the ambient temperature does not exceed 30ºC (86ºF).

5   Operation

Out of the box the m908 comes with a many factory presets that will allow for quickly connecting the most common input sources, cue, and speaker systems. If your installation requires customization, then see the Workflow Setup and Setup sections of this manual. Otherwise read this section for the basics on operating the m908.

The m908 uses a closed loop control system to ensure that the status reported on the RCU matches the actual hardware setting in the ACU. Any changes made to the configuration of the system are transmitted to the ACU, where they are processed and reported back to the RCU for confirmation. The following sections detail all of the controls and features of the normal operation mode.

5.1 Manual conventions

Throughout the Operation and Setup sections of this manual references to screen elements and setup menu parameters will use the following formatting:

- Hardware controls and connectors will be in ALL CAPITAL LETTERS
- Switches with secondary functions will be (in parentheses)
- Setup menu screen names will have the First Letter capitalized
- Setup parameter names will be in **bold lower case**
- Setup parameter values will be <enclosed>

- "push" refers to a quick push and release of a switch.
- "push-hold" refers to a 1/2 second push and hold of a switch.
- "push long-hold" refers to a 3 second push and hold of a switch.

5.2 Power Up

After making the appropriate power and signal connections turn on the m908 PSU.

NOTE: It is always recommended that power amplifiers and powered monitor speaker be turned on *after* the m908 to avoid any transients reaching the loudspeakers.

The illuminated switches on the RCU will go through a test sequence and then the LCD display will indicate that the system firmware is loading. This process takes approximately 1 minute.
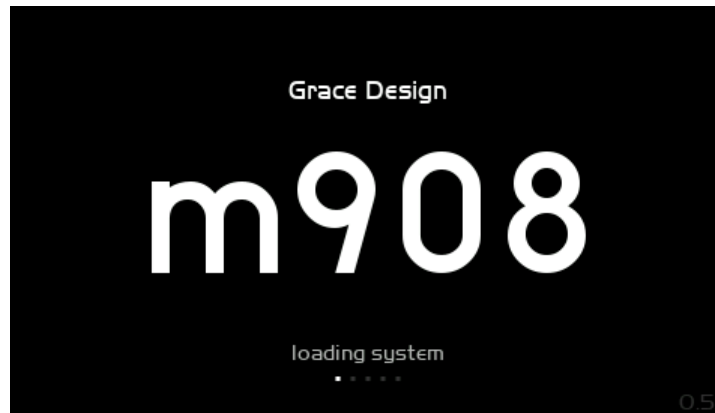
Illustration 9: m908 System Load Splash Screen

Once the system has loaded the RCU will display the Home Screen and the m908 will be operational.

5.3 Home Screen
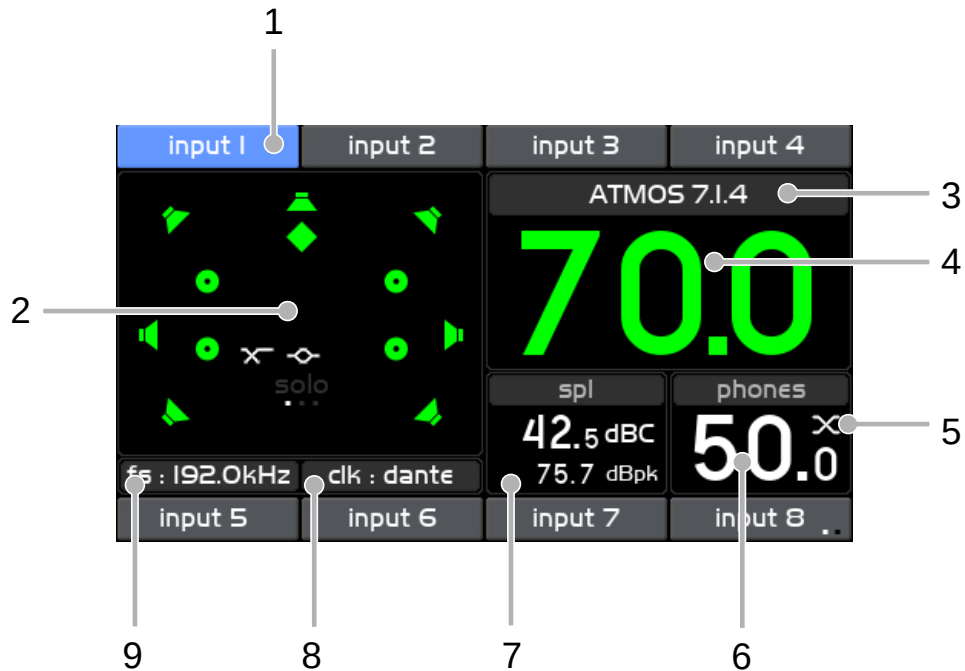
Illustration 10: Home Screen

The home screen provides an overview of the m908 system status.

5.3.1   1 Input Selection

The top and bottom bands of the LCD display area show the input sources available for monitoring. The currently selected input will be highlighted in blue. With Input Summing Mode active, the selected input(s) will be highlighted in green.

5.3.2   2 Control Room Speakers Layout Icons

The Control Room speaker system layout window displays a graphic representation of the currently selected speaker system. There are 4 types of speaker icons.

1. Front, Surround, and Back
2. Top
3. SUB
4. Height

Each icon also acts as a signal level indicator. The level is monitored after bass management, room EQ, and downmix processing and before speaker volume control and output delay..

- White No signal present.
- Green Signal greater than -60dBFS
- Yellow Signal greater than -12dBFS
- Red Signal 0dBFS (OVER)

In the lower center area of the speaker system layout window is the solo/mute page and status. The word "solo" will be visible in gray when the system is in solo mode. The word solo will be visible in white when there is a speaker in solo mode.

The word "mute" will be visible in gray when the system is in mute mode. The word "mute" will be visible in white when there is one or more speaker in mute mode.

Two icons indicate weather bass management and room eq are enabled:

Bass Management active:

Room EQ active:

The speaker system can be configured in the Speaker Output setup

5.3.3   3 Control Room Speaker Output

The name of the currently selected Control Room speaker system is displayed directly above the Control Room Monitor Level.

5.3.4   4 Control Room Monitor Level

The current Control Room monitor level is displayed in large numbers which will be green when the encoder is controlling the Control Room level. The default volume range is 0-100 in 0.5dB steps. 0 is mute and 96 is unity gain.

The Control Room Monitor Level can also be displayed with a user defined reference level. This way the volume readout can be calibrated to SPL.

See Monitor Control setup for details on configuring the Control Room Monitor Level display.

5.3.5   5 Headphone Crossfeed

The Headphone Crossfeed circuit status is displayed in the upper right corner of the headphones volume display window. See the X-FEED section of this manual for details.

5.3.6   6 Headphone Monitor Level

The current headphone monitor level is displayed in small numbers which will be green when the encoder is controlling the headphone level. If headphones are disabled in the system workflow then the Headphone Monitor Level display will read: HEADPHONES DISABLED. However, the headphone amplifier can still be used for low latency cue monitoring. See the CUE

setup section of this manual for details.

When enabled, the headphone amplifier signal source can be set to monitor the Left and Right channels of the current Control Room output, any of the CUE paths that are enabled, or the HP/CUE Downmixer.

See Monitor Control setup for details on configuring the Headphone Monitoring.

5.3.7   7 SPL Meter

Using the internal m908 RCU microphone, the m908 computes the real-time sound pressure level based on the system configuration. The current and peak SPL levels are displayed along with the filter mode in this section of the display. The peak spl value can be cleared at any time with a push-hold of the MON>CUE (spl-clear) switch. The SPL measurement settings are configured in SPL Meter setup.

When talkback is activated on the RCU, SPL monitoring will be suspended and the built in microphone will switch to become the talback mic. The SPL Display area will indicate TALKBACK ACTIVE. Upon exiting Talkback the SPL meter will return to normal operation.

5.3.8   8 Clock Source and Status

The clock status window displays the current clock source and indicates it's status by changing the color of the text. See the Clocking section of this manual for details.

5.3.9   9 System Sample Rate

The sample rate display indicates the sample rate detected on the input connector. If the input data or sample rate are invalid then the sample rate display will be blank (--).

5.4 INPUT SELECT

The m908 features a total of 16 input sources which are accessed with 8 dedicated INPUT Select switches, in 2 rows of 4, above and below the LCD display. Each INPUT Select switch is located adjacent to it's name area in the upper and lower bands of the display area. Pushing any of these switches will select the associated input source and connect it to the monitoring path. When an input is selected for monitoring it's name area background will be blue. Inactive inputs will have a gray background in the name area.

The 16 inputs are displayed across two "pages" of 8 inputs. To switch between the input pages push-hold the lower right INPUT select switch. The input name area next to the lower right hand INPUT switch will show which page is currently being displayed with two small dots. When the left dot is white inputs 1-8 are displayed. When the right dot is white inputs 9-16 are displayed.

Illustration 11: Input Page Select

Each input can be custom defined in the Setup Menu. The hardware connector, channel count, level offset, sync delay, clock override mode, and name can be configured. See the Input Setup section of this manual for details.

5.4.1   Input Summing Mode

The m908 allows up to three inputs to be summed. To enter summing mode, simultaneously press the INPUT select switches for up to three of the desired inputs to sum. The inputs will become active at the same time and the name areas will turn green on the display.
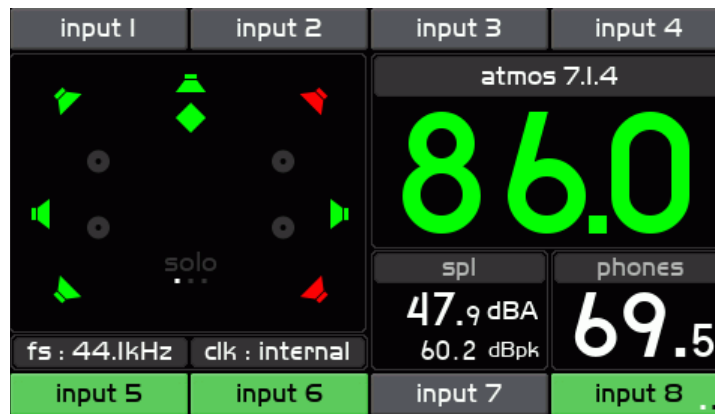
Illustration 12: Input Summing Mode

To change the inputs that are being summed press the INPUT select switches for the channels that are to be removed from the sum then press the INPUT select switches for the channels that are to be added to the sum.

Pressing any of the active INPUT select switches will toggle that input on and off in the sum.

To exit input summing mode simply de-select the inputs that you don't want in the sum. When only one input is active push that switch and summing mode will turn off. The input name area will be displayed in blue.

Any input that has **clock override** enabled can not be summed with other inputs unless all inputs share the same clock override source. Attempting to sum inputs that have different clock sources will cause the following message to appear:

Illustration 13: Input Summing Error: clocksource mismatch

Inputs that are not operating at the sample sample rate can not be summed. Attempting to sum inputs that are operating at different sample rates will cause the following message to appear:
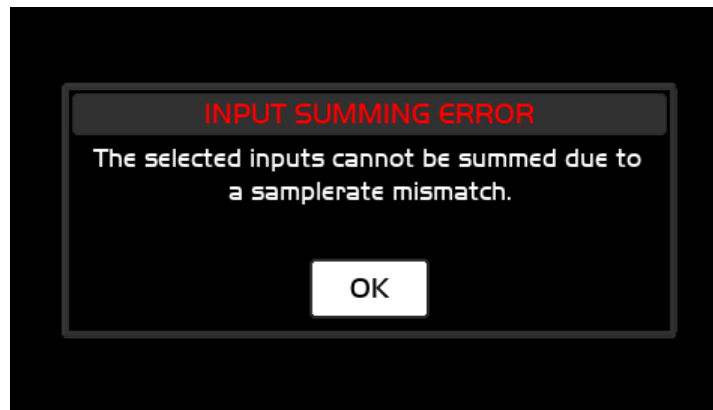
Illustration 14: Input Summing Error: sample rate mismatch

Note that inputs that are at the same sample rate but are *asynchronous* will be summed but there may be glitches due to clock drift.

Inputs that do not have identical Input **sync delay** settings can not be summed together. This is because the delay happens *after* the input summing section in the m908. Attempting to sum inputs with different sync delays will cause the following message to appear:
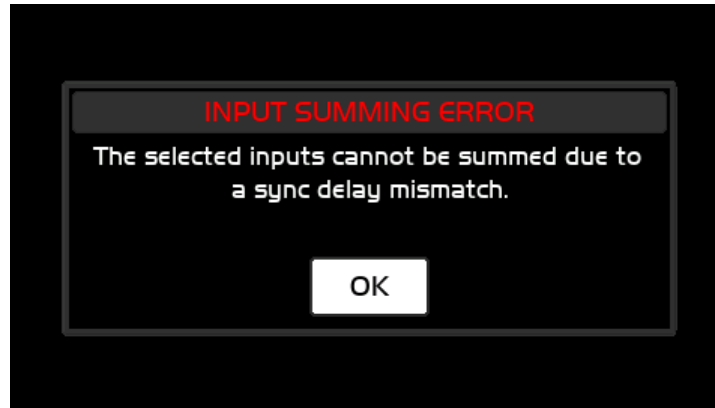
Illustration 15: Input Summing Error: sync delay mismatch

Since ADC1 channels 1 and 2 are shared with the balanced analog and unbalanced analog inputs, only one input can have these channels enabled in the summed input selection. Attempting to sum inputs that each contain ADC1 channels 1 and 2 will cause the following message to appear:
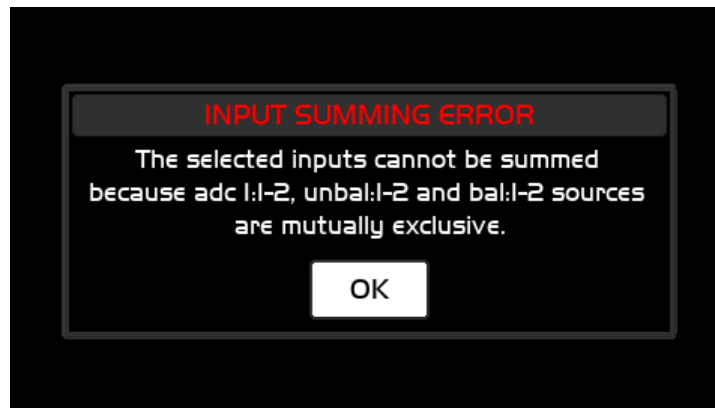
Illustration 16: Input Summing Error: ADC1 1-2 bal-unbal

Note: If the Meter Outputs are to provide metering for summed input signals the **meter output source** must be set for <follow monitor post> in System setup.

5.4.2   Direct Input Offset Adjustment

In some applications, the level of an input may need to be adjusted frequently. To avoid having use Input Select mode to make these changes, the m908 features a direct access input offset control.

While in the home screen, press and hold a single input button and turn the encoder to enter

the direct access input offset control input. Once engaged, the current input offset will be displayed in place of the modified input's name. This mode remains active until the inptut button is released or another input button is pressed simultaneously (which engages input summing mode).

 This feature is available for active inputs, inactive inputs and in summing mode. Changes made to input offsets via the direct access mode will not be saved in the workflow unless Setup mode is entered and then exited.

Illustration 17: Direct Input Offset on active Input
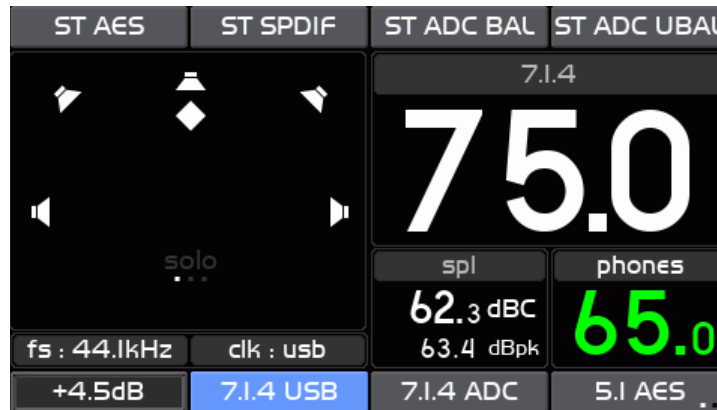
Illustration 18: Direct Input Offset on inactive Input
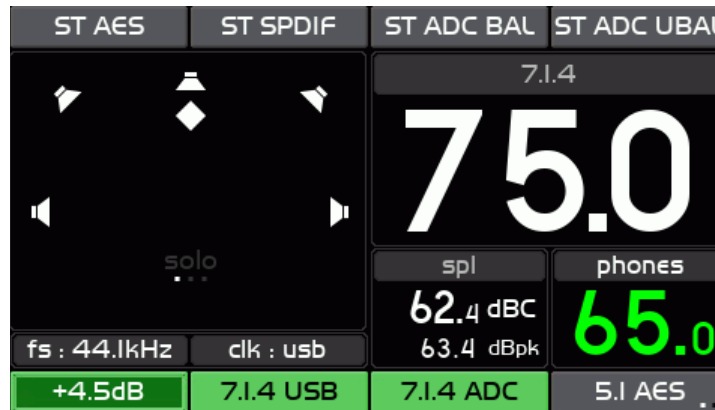
Illustration 19: Direct Input Offset on Summed Input

5.5 DIM

The multi-function DIM / (x-feed) switch controls the monitor DIM and headphone x-feed features.

Speaker DIM attenuates the current Control Room speaker system and/or the headphones by the preset attenuation level. When dim is active, the switch is lit solid yellow.

DIM level and which outputs it affects can be configured in Dim setup.

5.6 X-FEED

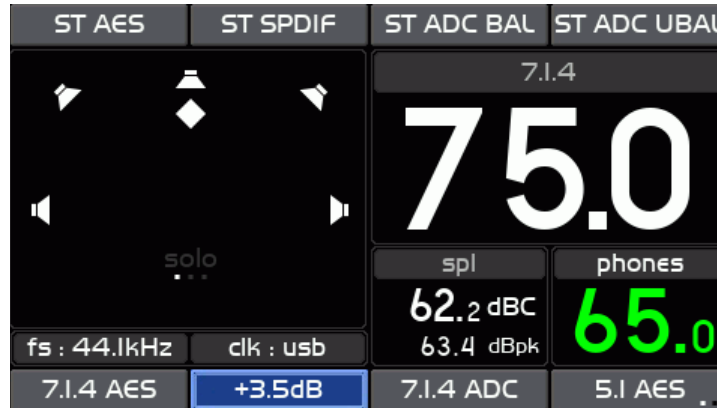Press-hold of the DIM / (x-feed) switch engages the Headphone cross-feed circuit in the headphone output. When x-feed is active it is indicated on the headphone status display. X-feed (cross-feed) simulates the acoustics of a loudspeaker listening environment which can significantly improve imaging while reducing listening fatigue when using headphones. This feature employs carefully designed signal cross-feed, filtering and delay circuits to simulate hrtf (head related transfer functions).

Press-hold the DIM (x-feed) switch to toggle Headphone Crossfeed on and off. Crossfeed is indicated in the upper right corner the phones volume window.

When listening to loudspeakers in a room, your left ear hears sound primarily from the left speaker (and vice versa) but also receives a signal from the right speaker at a lower level and with some time delay compared to the right ear. As well, the right speaker sound that reaches the left ear does not have a flat frequency response as the sound waves have traveled around the shape of your head before reaching your left ear. The brain uses delay, level and frequency response characteristics to process the location of a sound and hence, create an aural image.

When listening to a stereo (or down-mixed) signal through headphones, each ear only hears

the sound from one transducer and the mixing of signals between the ears does not exist. In this situation the brain is left without many of the psycho acoustic clues required to generate a properly distributed image and an accurate sound stage. The result is that sounds seem to cluster in the far left, far right or center of your head. Since the vital clues are absent, the brain has a difficult time deciding how to process the sounds coming from the headphone, which can result in listening fatigue when listening for extended periods of time. The m908 contains circuitry which electronically simulates the signal cross-feed that occurs in a real acoustic space and helps the brain establish instrument locations across the entire sound stage. While it is difficult to perfectly model the very complex level, delay and frequency response characteristics of the head, the cross-feed circuitry in the m908 gives the brain some of the basic clues it needs and the result is a very pleasing simulation of an acoustic space. We chose the parameters of the cross-feed circuit to find a good compromise between accurate imaging and tonal neutrality. For recreational listening there could be more aggressive modeling of the head related transfer function (HRTF) but this is usually at the expense of adding tone color. For critical monitoring during the recording/editing/mixing process the user will find that the m908 cross-feed circuit provides a sonically neutral character.

Below is a flow diagram and frequency response plot showing the response of the m908 crossfeed circuit. In this graph one channel of the headphone amplifier is driven. The two traces show the direct channel and the opposite (cross-feed) channel.
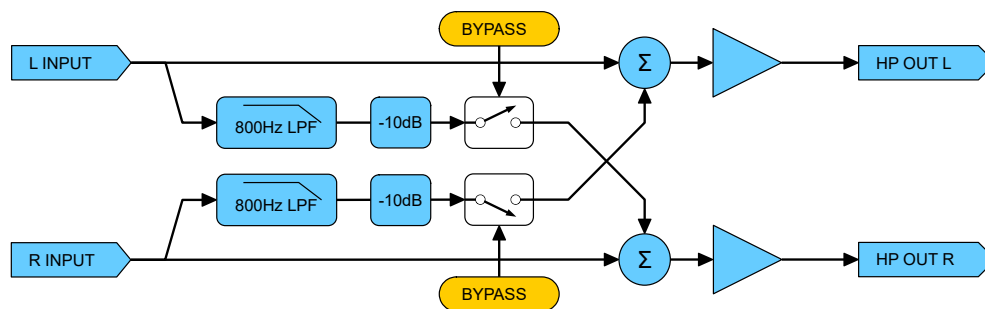
Illustration 20: Headphone Crossfeed flow diagram
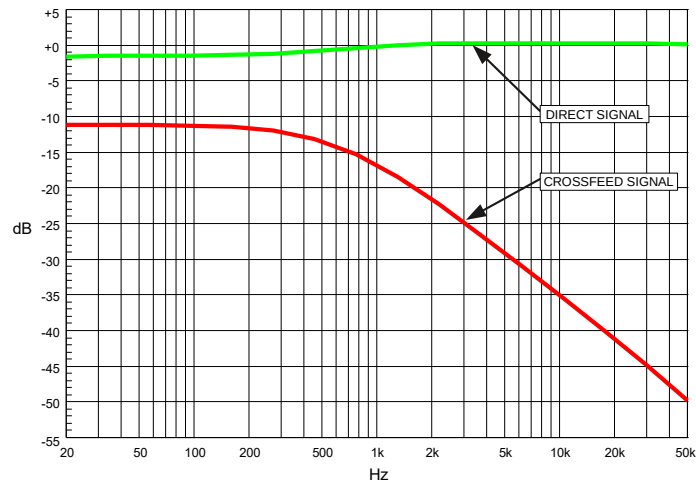
Illustration 21: Headphone Crossfeed Response Plot

5.7 MONO / (L-R)

The multi-function MONO / (L-R) switch controls the MONO and the L-R (Left minus Right) features.

Pressing the MONO causes the output of the m908 to play in mono.

When speaker / headphone MONO is active, the switch is lit solid yellow.

Press-hold of the MONO switch enters L-R mode. In this mode the m908 subtracts the right channel from the left channel and provides this signal to the monitor output.

When L-R is active, the switch flashes yellow.

Press and release MONO switch:

- Toggles mono on and off. Disables L-R if currently active.

Push-hold switch:

- Toggles the L- R mode on and off. Disables MONO if currently active.

The MONO and L-R processing can be applied to the Control Room speakers and/or the headphone outputs.

The signal source for the MONO and L-R processing can be the Left and Right channels of whatever input is being monitored. The output of the MONO and L-R processing can be the Left and Right speakers or the Center speaker.

When MONO or (L-R) are enabled on a multi-channel CR Output all other channels continue to play normally.

See the MONO setup and (L-R) setup for configuration options.

5.8 MON>CUE

The MON>CUE switch activates Monitor to Cue mode which routes the currently selected Control Room monitor source to the CUE outputs. The MON>CUE function can be enabled or disabled for each of the cue outputs. Also, the send to the cue outputs can be the Left and Right channels of the current monitor source or a stereo downmix if the source is in surround.

When active the MON>CUE switch will illuminate green.

See the [CUE setup](#) for configuration options.

5.9 MUTE

The MUTE switch mutes the audio in the Control Room speaker and/or the Headphones. When active the MUTE switch will illuminate red.

See the [MUTE setup](#) for configuration options.

5.10 CR1, CR2, CR3

Speaker system selection is made with one of the three CR (Control Room) switches.

Each output is defined in the Setup Menu. The hardware connector, channel count, level offset, delay, bass management parameters, room correction eq, and name can be configured in the [CR Setup](#) section of this manual.

The Control Room speaker select switches are exclusive and only one can be selected at a time. The currently selected speaker switch will illuminate green.

5.11 BASS MANAGEMENT

The m908 includes comprehensive Bass Management control. Each speaker in the Control Room system can be individually configured for crossover frequency, high pass filter slope, and low pass filter slope. The internal summing of the LPF signals is done in stereo. All left channel speakers are added to the Left sub bus, right channel speakers are added to the Right sub bus, and center speakers are added to both buses equally. The user can decide to use stereo sub woofers or use a mono sum of the stereo bus. There are three dedicated sub woofer output channels.

The high pass and low pass filter slope options are:

- 24dB/octave Linkwitz-Riley
- 18dB/octave Butterworth
- 12dB/octave Linkwitz Riley (LPF Output inverted)
- 6dB/octave

The high pass filter can be bypassed so a full range signal is sent to the speaker. As well, the low pass filter can be turned off so that no signal is sent to the sub woofers.

Sub mode options allow for a variety of sub configurations. A sub output can contain a mix of bass managed signals from the LPFs with or without the LFE as well as LFE only.

The crossover filter and sub mode settings can be found in the Bass Management setup screen.
The Bass Management system can be bypassed from the Monitor Control setup screen.

The LFE input channel contains a variable low pass filter and a 0 or +10dB gain setting. These are system level settings which are contained in the Monitor Control setup.
See the Monitor Control setup section of this manual for configuration details.

CAUTION: When enabling or disabling Bass Management in Monitor Control Setup it is recommended that you turn your Control Room volume down to 0.

IN 2,3...   25-150Hz HPF   WORKFLOW DEPENDENT   OUT 2, 3...

25-150Hz LPF

IN 1   25-150Hz HPF 6,12,18,24dB/oct or BYPASS   OUT 1

WORKFLOW DEPENDENT

25-150Hz LPF 6,12,18,24dB/oct or OFF

LEFT SUB BUS   RIGHT SUB BUS

Σ

SUB MODE

IN LFE   80-120Hz LPF   0dB +10dB
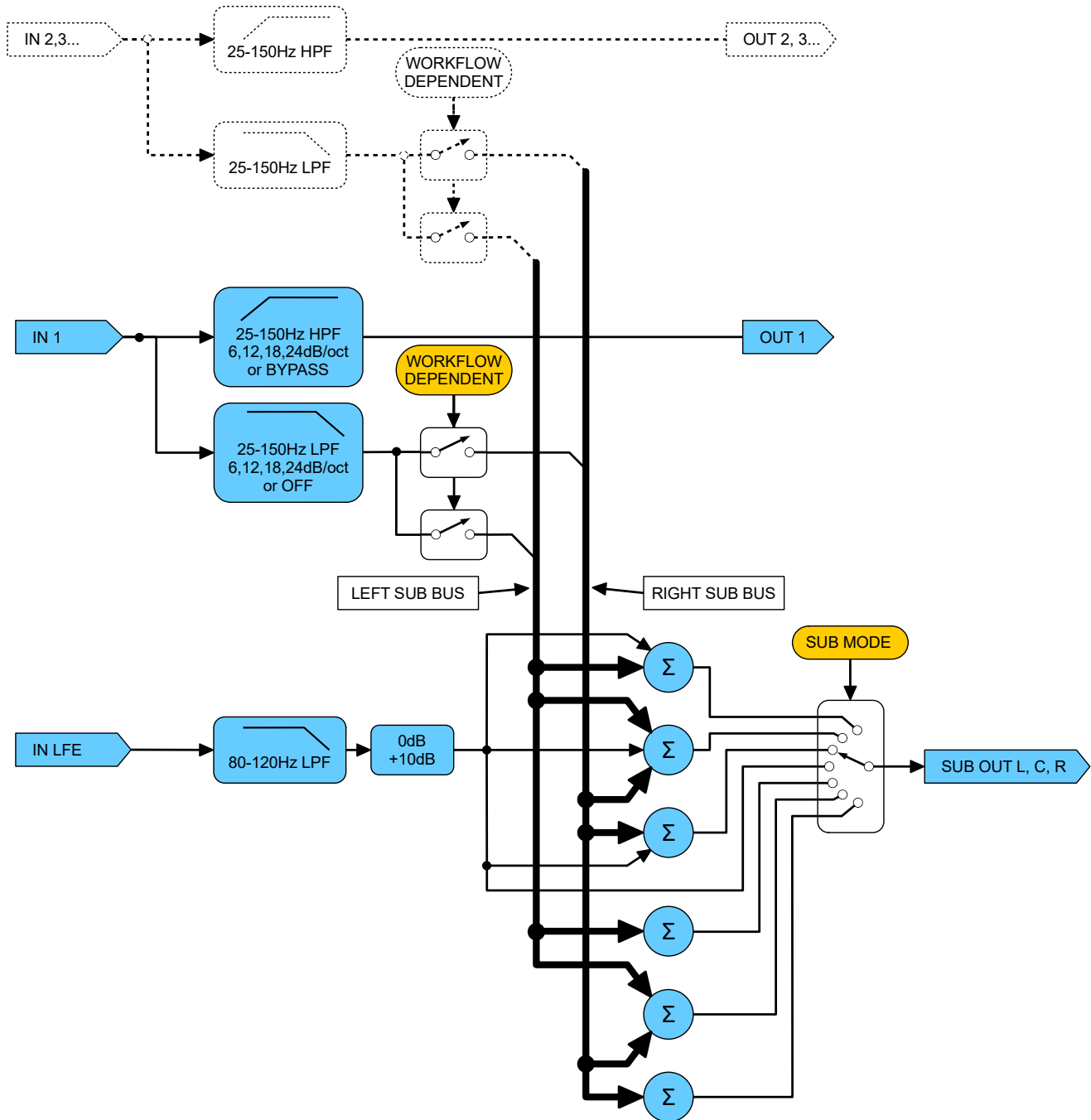
Σ

Σ   SUB OUT L, C, R

Σ

Σ

Σ

Illustration 22: Bass Management Flow Diagram

5.12 ROOM CORRECTION

The Control Room speaker output channels each contain up to 12 fully parametric bands of equalization for correcting minor frequency response aberrations in your control room.

Each band of eq can be set for high pass, low pass, high shelf, low shelf, and peak response. All room correction equalizers can be bypassed from the Monitor Control setup screen. See the Room Correction setup section of this manual for details.

5.13 CUE

The m908 can be configured to have up to 8 stereo cue paths. These cue paths can be assigned to any hardware input and output connector except UNBAL, BAL, and ADC1 1-2 in, CR1,CR2, and CR3 out. They are intended for allowing cue mixes to be sent to the studio while allowing for Talkback to be mixed in for communication with studio musicians/talent. The cue routing system is designed for very low latency for monitoring while recording.

Pushing the MON>CUE switch routes the current Control Room monitor source to the Cue output. The MON>CUE output select setup parameter allows each Cue output to send the Left/Right CR Output, downmix, or nothing when MON>CUE is active. See the Cue setup section of this manual for details.

If a Cue output is set to receive a downmix of the Control Room Monitor source the type of downmixer used is determined by the **phones/cue downmix mode** setup parameter in Monitor Control setup. The available downmix templates are identical to the downmix templates for the Control Room outputs but are stereo output only.

The Cue input and output connector routing is configured in CUE setup. Note that both left and right channels of a cue *input* path must be assigned to the same connector.

The Headphone amplifier can be set to output any one of the Cue outputs. This is convenient if an external headphone amplifier is not available. See the Monitor Control setup section of this manual for details.

There are two ways that a Cue signal can be routed to the Headphone amplifier:

- Assign the Cue output to CUE/CR EXT. channels 7-8. In this mode the Cue output signal will be sent to the CUE/CR EXT channels 7 and 8 as well as the Headphone amplifier Left and Right channels. Level control is done in the CUE setup. For this to work the Headphones must be *disabled* in the Workflow. This mode provides headphone monitoring of the cue signals with minimum latency (<1ms) but headphone processing for MONO, MUTE, DIM, and L-R are not available.
- Assign the Headphone Amplifier signal source to the desired Cue path with the **headphone source** parameter in Monitor Control setup. It is not necessary to assign

routes to the Cue output. Level control is done using the main Volume control encoder on the RCU. For this mode Headphones must be enabled in the Workflow. This mode provides headphone monitoring of the cue signal and maintains RCU VOLUME control, MONO, MUTE, DIM, and L-R processing. However, there will be 4ms of latency from Cue in to headphones out.

If a Cue input is assigned to a digital input connector the input signal must be at the same sample rate and synchronous with the System Clock Source. If any Cue input is at a different sample rate from the System Clock Source it's *input* signal will be muted. However, TALKBACK and MON/CUE will remain active. When any of the Cue paths are muted due to sample rate mismatch "cue ip mute" will illuminate in the Speaker Layout area of the RCU display.
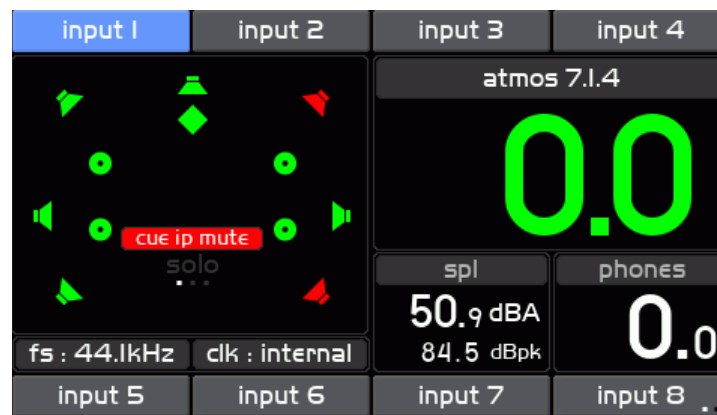

Illustration 23: Cue Input Mute Indication

To avoid muting Cue inputs on sample rate mismatches it is recommended to assign Cue inputs to ADC input channels.

5.14 TALKBACK

The m908 has a flexible talkback system that can be adapted to a wide range of applications. There is a microphone built in to the RCU and a talkback mic amp built in to the ACU. The talkback microphones can be controlled from multiple trigger sources and routed to multiple CUE paths. Trigger sources include the RCU TALKBACK switch, an external talkback switch jack on the ACU, A B and C User switches, and GPIO input pins. As well, GPIO pins can be programmed to output a logic level tally when talkback is active. Multiple routing configurations allow different talkback triggers to route talkback audio to various CUE sends. For instance, you can have the main TALKBACK switch on the RCU send talkback audio to all cue sends while USER A switch can send talkback audio to only a specific CUE output.

The analog XLR Talkback Output jack on the ACU outputs the talkback signal from the

currently active talkback mic. This allows the Talkback signal to be sent to other headphone systems or studio playback speakers.
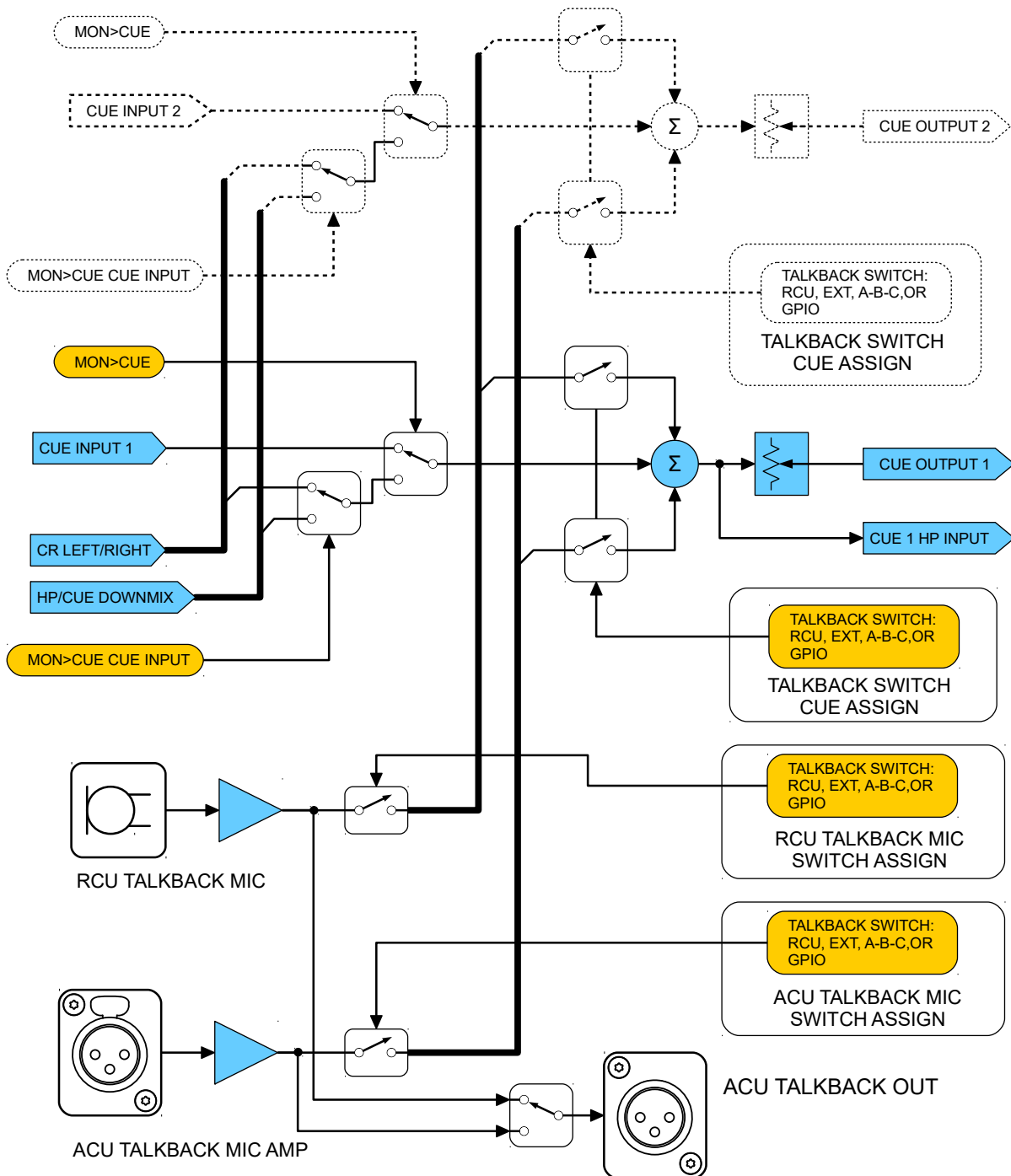
Illustration 24: Talkback-Cue Flow Diagram

The TALKBACK, ABC, and external talkback switches use an automatic latch/momentary operation. A quick press-release will latch the talback mic on. Press-release again to turn talkback off. Pushing and holding the talkback switch will activate talkback only while the switch is being held down (momentary mode). When Talkback is activated by pressing the TALKBACK switch it will illuminate red. If Talkback has been activated by the ABC switches, External Talkback switch or a GPIO pin, the TALKBACK switch will flash RED.

See the Talkback setup section of this manual for configuration details
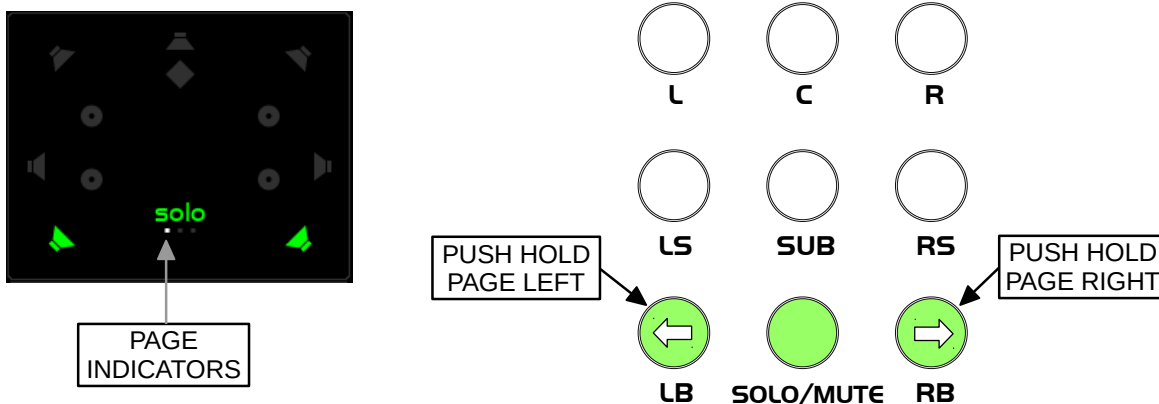
5.15 SOLO/MUTE

The nine bi-colored SOLO/MUTE switches are used to solo or mute any individual channel or group of channels being monitored. Eight of the switches engage solo or mute for the relevant channel, while the solo/mute switch is used to toggle between solo or mute mode.

When in Solo mode the SOLO/MUTE switch will illuminate green. Pushing any of the 8 SOLO/MUTE switches will toggle solo mode on and off for that channel. In the speaker layout window the word "solo" will appear in green and channels that are not soloed will be grayed out.

When in Mute mode the SOLO/MUTE switch will illuminate red. Pushing any of the 8 SOLO/MUTE switches will toggle mute mode on and off for that channel. In the speaker layout window the word "mute" will appear in red and channels that are muted will be grayed out.

The SOLO/MUTE switches handle a complete 7.1 speaker system. However, if your speaker system has more than 8 channels the SOLO/MUTE switches can "page" to the higher order channels.

To page to higher order channels push-hold the RB switch. To page to lower order channels push-hold the LB switch. The current solo/mute page is indicated in the group of 3 small dots in the middle area of the speaker layout section of the LCD display. The current solo or mute mode

Illustration 25: Solo/Mute Page Keys

With the three "pages" of SOLO/MUTE switches there are a total of 24 controls. From the factory the first 8 of the SOLO/MUTE switches are mapped to speaker channels as indicated on the RCU panel: L, C, R, LS, SUB, RS, LB, RB. SOLO/MUTE switches 9-24 (pages 2-3) are unassigned.

More than one speaker channel can be mapped to a SOLO/MUTE switch. This allows for solo/mute "groups" to be created. For instance, you could have a SOLO/MUTE switch assigned to all of the overhead speakers in an ATMOS system. One button press would then allow muting and soloing of the whole group of speakers at once.

The mapping of speaker channels to SOLO/MUTE switches can be changed in the SOLO MUTE Setup section of this manual.

5.16 VOLUME (push for phones)

The m908 utilizes a single rotary push-button encoder for the Control Room speaker volume level, Headphone volume level, and volume level preset recall.

Pushing the VOLUME encoder toggles between Control Room and Headphone control.

Turning the VOLUME encoder increases and decreases the monitoring level in 0.5dB steps.

The VOLUME control can also be set to coarse/fine mode where each volume step is 4.5dB. In this mode pushing and turning the encoder knob will make 0.5dB steps.

The volume control is speed sensitive, so the turning the knob at a higher velocity results in faster volume changes with larger step sizes.

With Control Room Speaker control active:

Push-hold of the VOLUME encoder will set the monitoring level to the Preset Level.

There are two modes of level display for the Control Room level:

- Standard Mode The Control Room level range is 0-100 with 0 being mute and 96 being unity gain. With 0.5dB steps this is a 99.5dB attenuation range.
- Reference Mode The Control Room level readout can be scaled so that a calibrated monitoring level can be displayed as 0dB.

With Headphone control active:

Push-hold of the VOLUME encoder will set the monitoring level to the Preset Level.

See the Monitor Control setup section of this manual for configuration details.

5.17 A B C User Switches

The ABC User Switches are software definable and can be configured to control a variety of system commands.

Available uses are additional talkback triggers, downmix control, and GPIO triggers.

See the A B C User Switches Setup section of this manual for configuration details.

When configured for Talkback the User switches will behave just like the dedicated Talkback

switch and will illuminate RED when active.

When configured to be GPIO triggers the User switches will toggle on and off and illuminate GREEN when active.

5.18 Meter Outputs

The m908 allows for flexible routing of input signals to Meter Outputs. In addition to providing output for metering the routing facilities allow for alternate audio paths as well. For instance, an unused ADC input can be routed to an unused digital output connector to allow for stand alone ADC operation.
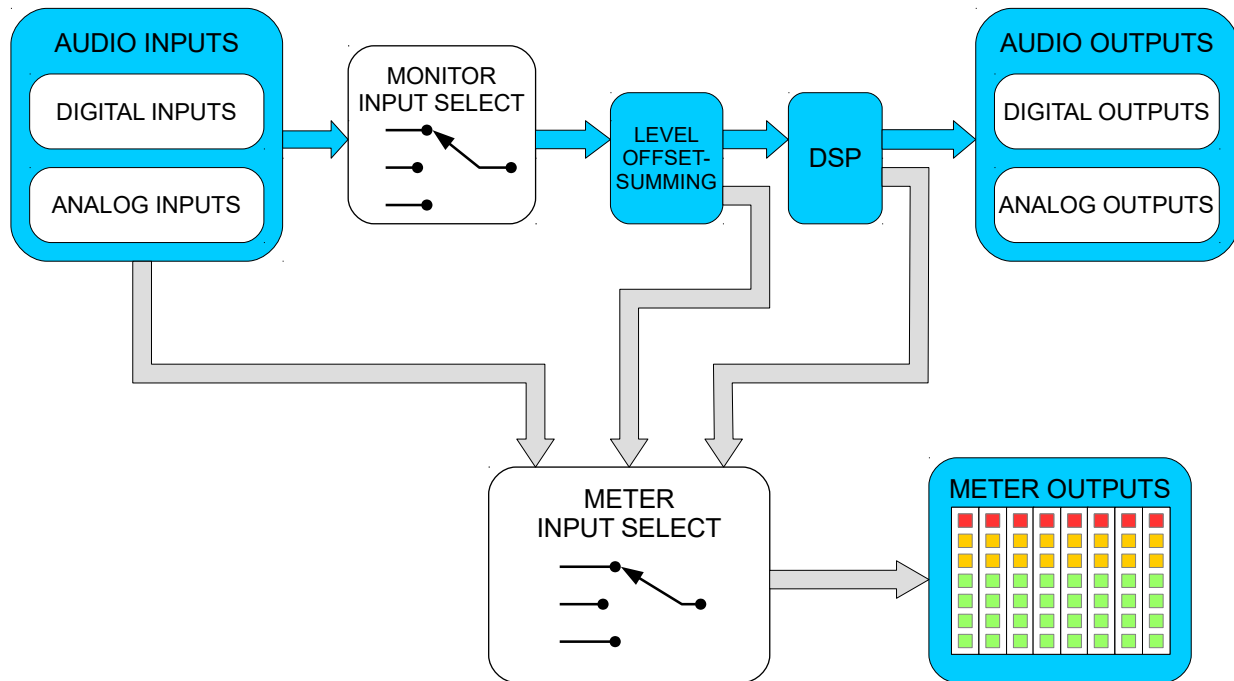

Illustration 26: Meter Output Routing

The meter source options are:
- Audio Input source
    This option fixes the Meter source to a specific input. The signal level is identical to the

incoming signal level.

- Follow monitoring source selector pre DSP

    This option allows the Meter source to follow whatever is being monitored on the Control Room speakers. The signal level follows the Input Level Offset and Input Summing.

- Follow Monitor Post DSP output.

    This option sources the meter output from the final outputs for the Control Room speakers. The meter signal will have all speaker processing applied: volume, delay, offset, bass management, and room EQ.

The Meter outputs can be assigned to any hardware output connector that is not being used by a Control Room speaker or a Cue output.

Note: CR1, CR2, and CR3 OUT can not be used for meter outputs.

See the Meter Output setup section of this manual for details on Meter Output routing.

5.19 Downmix

The m908 contains two downmixers and a predefined set of downmix templates. The first downmixer processes the input signal and outputs to the CR Outputs. The second downmixer processes the input signals and is available as a source to feed the Headphone and Cue outputs. This downmixer uses only stereo out templates and allows these stereo Cue and Headphone outputs to monitor a surround input source while simultaneously monitoring in surround on the CR Outputs. Each stereo output template contains downmix level parameters. These parameters are set to default Dolby metadata values in most cases. Downmix templates are accessed by configuring one of the A,B, or C User switches to activate a downmix.

To enable a downmix mixer one of the A, B, or C User switches must be assigned to downmix mode with a downmix template selected. See the A B C User Switches Setup section of this manual for details.

Up to three downmix templates can be used by assigning each A, B, and C switches to a downmix template.

The CR Output downmix templates can be activated by assigning one of the ABC switches to the downmix function. See the A B C User Switches Setup section of this manual for details.

For Downmix to work properly the Input Source and the Control Room output speaker system must match the input and output channel configuration of the selected Downmix template. For instance, if a 7.1 downmix template is enabled, the Input source should be 7.1 and the Control Room output speakers should have at least as many channels as the downmix output. Also, the Workflow must contain a speaker layout using the names below. The INPUT and CR output must be routed to these speaker names.

Speaker Channels for Downmix Operation	
Speaker Name	**Abbreviation**
LEFT	L
CENTER	C
RIGHT	R
LEFT MIDDLE SURR	LMS
RIGHT MIDDLE SURR	RMS
LEFT BACK	LB
RIGHT BACK	RB
CENTER BACK	CB

Illustration 27: Downmix Template Speaker Layouts

The downmix templates use the following summing formulas
- 7.1 > 5.1
  - LMS = LMS + LB
  - RMS = RMS + RB
- 7.1 > 5.1 PLII (Pro Logic II)

- ○ LMS = LMS + (–1.2dB x LB) + (–6.2dB x RB)
- ○ RMS = RMS + (–6.2dB x LB) + (–1.2dB x RB)
- 7.1 to LCRS
  - ○ CB = (-3dB x (LMS + LB)) +(-3dB x (RMS + RB))
- 7.1 to LCRS2
  - ○ LMS = (-6dB x (LMS + LB)) +(-6dB x (RMS + RB))
  - ○ RMS = (-6dB x (LMS + LB)) +(-6dB x (RMS + RB))
- 7.1 > Lt/Rt
  - ○ L = L + (–3 dB x C) – (–3 dB x (LMS + LB)) – (–3 dB x (RMS + RB))
  - ○ R = R + (–3 dB x C) + (–3 dB x (LMS + LB)) + (–3 dB x (RMS + RB))
- 7.1 > Lt/Rt PLII (Pro Logic II)
  - ○ L = L + (–3 dB x C) – (–1.2 dB x (LMS + LB)) – (–6.2 dB x (RMS + RB))
  - ○ R = R + (–3 dB x C) + (–6.2 dB x (LMS + LB)) + (–1.2dB x (RMS + RB))
- 7.1 > Lo/Ro
  - ○ L = L + (–3 dB x C) + (–3 dB x (LMS + LB))
  - ○ R = R + (–3 dB x C) + (–3 dB x (RMS + RB))
- 5.1 to LCRS
  - ○ CB = (-3dB x LMS) + (-3dB x RMS)
- 5.1 to LCRS2
  - ○ LMS = (-6dB x LMS) + (-6dB x RMS)
  - ○ RMS = (-6dB x LMS) + (-6dB x RMS)
- 5.1 to Lt/Rt:
  - ○ L = L + (–3 dB x C) – (–3 dB x LMS) – (–3 dB x RMS)
  - ○ R = R + (–3 dB x C) + (–3 dB x LMS) + (–3 dB x RMS)
- 5.1 to Lt/Rt (Pro Logic II)
  - ○ L = L + (–3 dB x C) – (–1.2 dB x LMS) – (–6.2 dB x RMS)
  - ○ R = R + (–3 dB x C) + (–6.2 dB x LMS) + (–1.2 dB x RMS)
- 5.1 to Lo/Ro
  - ○ L = L + (–3 dB x C) + (–3 dB x LMS)
  - ○ R = R + (–3 dB x C) + (–3 dB x RMS)
- LCRS to L/R
  - ○ L = L + (–3 dB x C) + (–3 dB x CB)
  - ○ R = R + (–3 dB x C) + (–3 dB x CB)
- LCRS2 to L/R
  - ○ L = L + (–3 dB x C) + (–3 dB x LMS)

- R = R + (–3 dB x C) + (–3 dB x RMS)

For downmix templates that have stereo output the Center channel summing level can be set to the following values:

Center Downmix Level
+3.0dB (1.414)
+1.5dB (1.189)
0dB (1.000)
-1.5dB (0.841)
-3.0dB (0.707) default
-4.5dB (0.596)
-6.0dB (0.500)

For downmix templates that have stereo output the surround channel summing level can be set to the following values: Note: ProLogic II surround channel summing level is not adjustable.

Surround Downmix Level
-1.5dB (0.841)
-3.0dB (0.707) default
-4.5dB (0.596)
-6.0dB (0.500)

See the Downmix setup section of this manual for configurations details.

5.20 Headphone Source Select

The Headphones, if enabled in the Workflow Setup, can be sourced from several signal sources. For normal monitoring of what is being played in the Control Room, the headphones can be sourced from the CR Left and Right channels. If a surround source is being monitored in the Control Room then the headphones can be sourced from a separated downmixer that will output a stereo version of surround signal. As well, the headphones can be assigned to a Cue path. This allows the headphone amplifier to be used for Cue monitoring if an external headphone amplifier is not available. See the Monitor Control setup section of the manual for details.

Note that if a Cue path is being monitored the input and output connector routing for the Cue path is still active so the headphones and whatever output connector has been defined for the Cue

path will both be active. The Cue output level control is independent of the headphone volume. See the CUE setup section of this manual for details.

The headphone amplifier signal is wired to the DAC 2 channels 7-8. These are also wired to the CUE/CR EXT. DB25 connector on the rear panel of the ACU. When headphones are active in the workflow no signal will be present on CUE/CR EXT. channels 7-8.

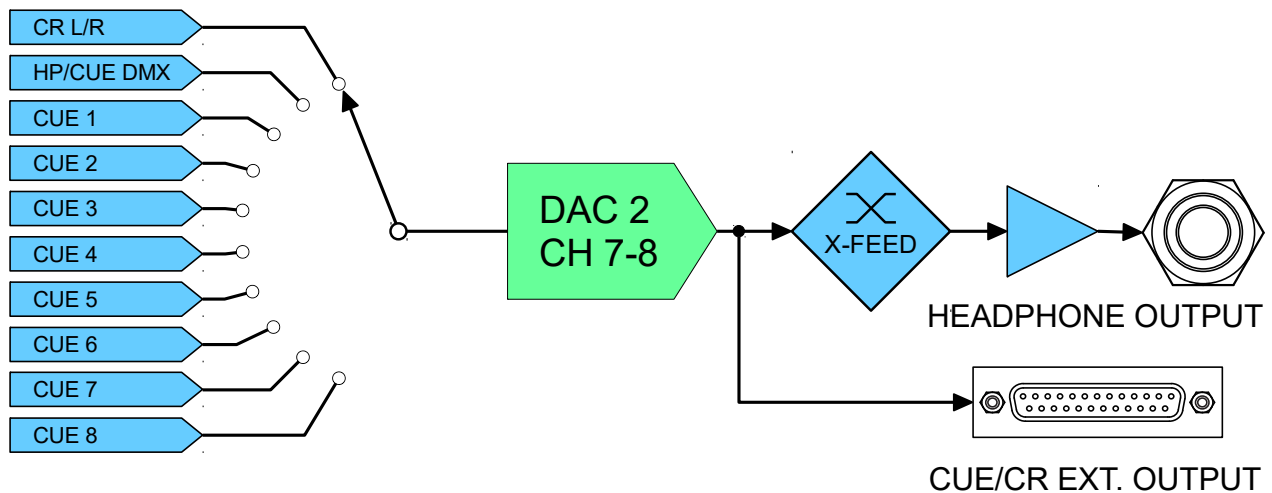The illustration below shows the headphone selection.

Illustration 28: Headphone Source Select

5.21 Clocking

The m908 System Clock can run from an internal crystal reference or it can lock to external Word Clock or digital audio input. See Clock Source Setup for configuration details.

The m908 will attempt to lock to the clock source selected in System setup. If the selected clock is at a valid sample rate (see Electrical Specifications for frequency ranges) the m908 will lock and the Clock Display will indicate the lock status.

If the System Clock is present but not at a valid sample rate the m908 will attempt to lock to the audio data source instead and play audio.

If the input source is composed of more than one hardware input source (4 AES3 signals for instance) the m908 will lock to the lowest channel number. All signals are assumed to be at the same sample rate and synchronous to each other. If there are mismatched sample rates among the input signals for the input source then audio will be muted and the system clock status will flash

red.

The m908 does not contain a sample rate converter. If the clock source is set to a connector that is not the same as the audio data source then it is assumed that the audio data will be at the same sample rate and be synchronous to the clock source.

If the m908 system clock source is set to one of the Internal sample rates then the m908 is the clock master. Any digital audio source connected to the m908 will need to be synchronized to the m908 word clock output.

If there is a digital input source that can not be synchronized to an external clock (CD player for instance) then set the **override clock** in the Input Setup for that device. When that input is selected for monitoring, the System Clock will switch the clock source to that input audio stream automatically. See the Input Setup section of this manual for details.

Below are two tables that summarize the various clock and sample rate states that are indicated by the fs: and clk: displays on the RCU.

Clock Status			
Clock Status Display	System Clock Status	Audio Source Condition	User Action Required
Flashing-White	Indicated clock source is valid but sample sample rate frequency is out of the PLL lock range.	m908 will play audio, but playback may have pops/clicks	Reconfigure input for samplerate in the valid lock range
White	Locked to indicated System Clock source	m908 will play audio using indicated clock.	None
Flashing Yellow	Indicated clocksource is invalid or does not match the samplerate of the active input. Active input clock is valid for digital inputs or system clock is valid for analog only inputs.	m908 plays audio with active input clock for digital inputs or system clocksource for analog only inputs	For inputs asynchronous to system clocksource, configure input clock override.\n\nFix invalid clock override\n\nConfigure system for a clocksource that is synchrous to input
Flashing Red	Indicated clocksource is invalid or does not match the samplerate of the active input. Active input clock is invalid.	m908 cannot play audio and outputs are muted	For inputs asynchronous to system clocksource, configure input clock override.\n\nFix invalid clock override\n\nConfigure system for clocksource synchrous to input\n\nRefer to fs display for required input action

Sample Rate Status			
Sample Rate Display	System Clock Status	Audio Source Condition	User Action Required
44.1kHz – 192kHz (White)	Current Sample Rate	m908 plays audio	None
--- (White)	With analog only input active: invalid system samplerate With digital input active: input signal unlocked	m908 cannot play audio and outputs are muted	Analog input(s): set system clocksource to a valid clock Digital input(s): connect a valid source"
fs mismatch (Flashing Red)	Digital input has signals at multiple sample rates	m908 cannot play audio and outputs are muted	All digital signals in a user defined input must be synchronous

5.22 Word Clock In and Out

The m908 can accept a standard 5V/75Ω word clock signal from an external clock generating unit. This might be a stand-alone master clock source or the word clock output from your digital audio workstation interface. The m908 locks to the incoming word clock with an ultra low jitter s-Lock PLL. The PLL has a fast lock mode which rapidly acquires lock and then switches to a high jitter rejection mode with a loop bandwidth of 0.5Hz. This provides exceptional jitter rejection for the ADC and DAC sample clocks. In the event of a dropout or loss of incoming word clock signal, the intelligent PLL will remain at the last known valid frequency. When the signal is restored the PLL will smoothly re-acquire lock.

The word clock output allows the user to synchronize other digital audio equipment to the m908. When the menu setting **clock output** is set to <clock in> this connector output is a copy of the signal on the Word Clock IN jack. This mode is useful for "daisy chaining" multiple devices together in a system. When **clock output** is set to <system clock> this connector outputs a copy of the m908 system word clock. The Word Clock output jack is buffered and is designed to drive a 75Ω line at 5V.
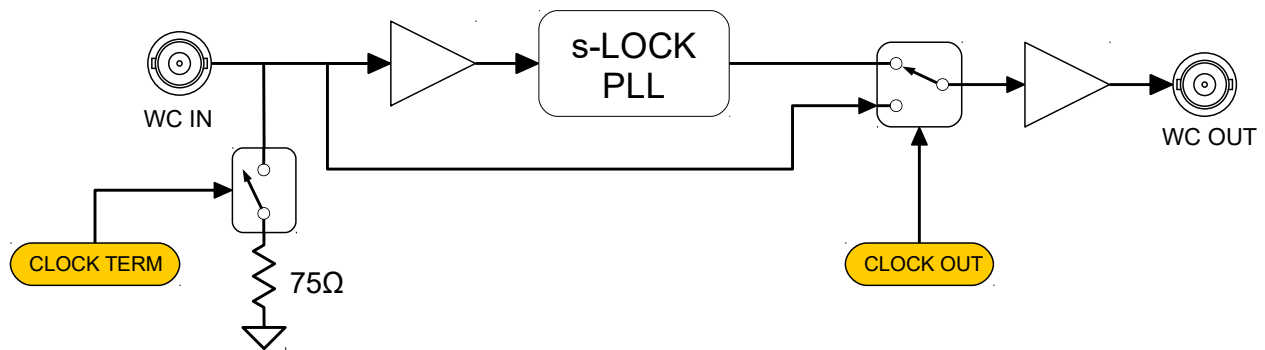
Illustration 29: Word Clock Flow Diagram

If the **clock output** parameter is set for <system clock> and the m908 is being synchronized to an external word clock, the Word Clock output will be a re-clocked low jitter version of the incoming clock. If the incoming clock is interrupted the m908 will continue to transmit on the Word Clock output at the last frequency that the system was locked to. When the input clock is restored the m908 will re-lock smoothly unless then new incoming sample rate has changed.

See Word Clock Termination Setup and Word Clock Output Setup for setup instructions.

The word clock polarity is high for left channel data and low for right channel data.

5.23 Error Messages

The ACU monitors several hardware systems for fault conditions and will report errors if there are any detected faults.

5.23.1 ACU Over Temperature

The ACU contains a temperature sensor on the circuit board near the main DSP processor. If the temperature rises above a safe level for the processor then the RCU will report an over temperature condition.

If this message appears check that the ACU has proper ventilation and that the ambient temperature where it is installed is below 40°C (104°F). Verify that the fan exhaust on the right side of the chassis and the cool air inlet openings on the left side of the chassis are not obstructed.

Illustration 30: ACU Temperature Error

5.23.2 PSU Error

The PSU contains a processor that monitors the voltages from the two redundant power supply modules. If any of these voltages fall outside of the permitted tolerance then this will be communicated to the ACU and a warning will appear on the RCU display screen. If this error occurs contact Grace Design Technical Support
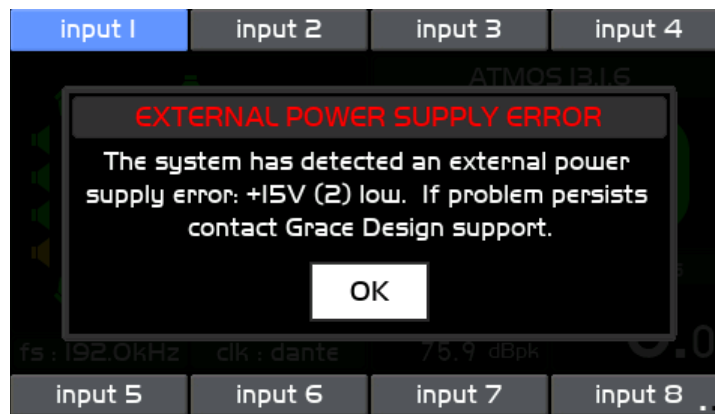

Illustration 31: PSU Error

5.23.3 Communication Error

The RCU and ACU communicate over a balanced RS485 serial link. If the RCU experiences a communication error this screen will be displayed.
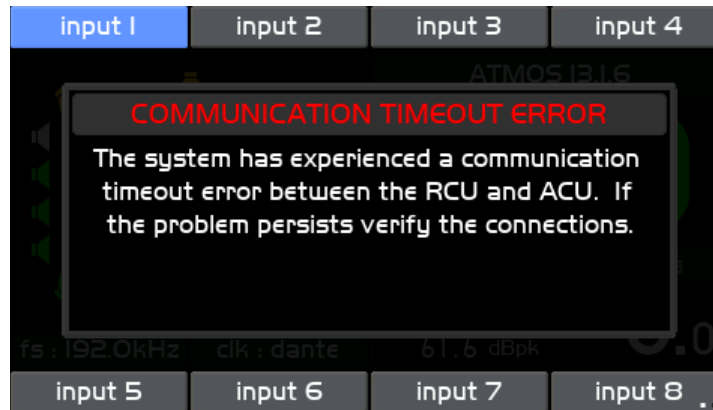
Illustration 32: Communication Error

5.23.4   Cooling Fan Error

The m908 contains a cooling fan that will turn on if the internal temperature of the m908 rises above a preset level. At power up the ACU processor tests the fan to ensure that it is runs at the proper rpm. If the fan fails to rotate at the proper rpm this error message is shown.
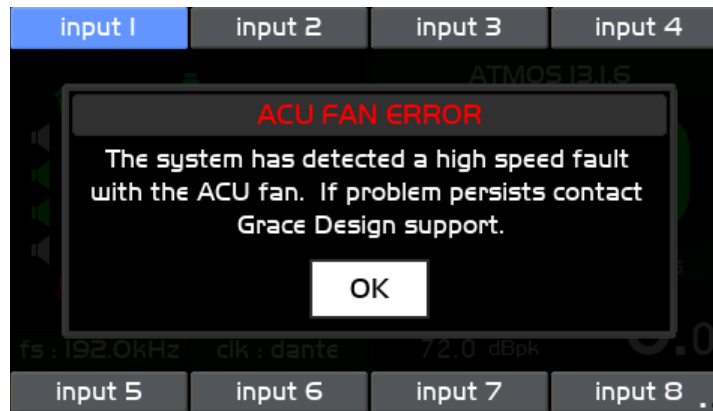

Illustration 33: Cooling Fan Error

6   System Setup

6.1 Introduction

There are two setup menus for the m908: *Workflow Setup*, and *System Setup*. Workflows are system presets that determine the mode that the DSP section of the m908 will operate in.

Workflow Setup allocates the 24 DSP channels for the speaker system, cue system, stereo headphone monitor, and talkback system. Speaker system formats range from Stereo to 24

GRACE DESIGN

channel monitoring. When choosing a speaker system format that uses less than 24 channels, the remaining processor channels can be assigned as cue channels, stereo headphone monitor, and talkback.

For instance, if you select a 7.1 Workflow, 8 channels of the processor will be allocated to speaker processing, while the remaining 16 channels can be allocated to 4 stereo Cue channels, stereo headphones, and two talkback mics.

Note that the number of channels assigned to the speaker system represents the *maximum* number of speakers that can be used. However, the m908 supports multiple speaker systems so, for instance, with the example above you could have a 7.1, 5.1 and stereo speaker system connected to the m908.

In the System Setup menu the input sources and speaker outputs can be configured. Input sources can be assigned to hardware connectors and routed to the appropriate DSP channels. Speaker systems can be assigned to hardware output connectors and routed from the appropriate DSP channels.
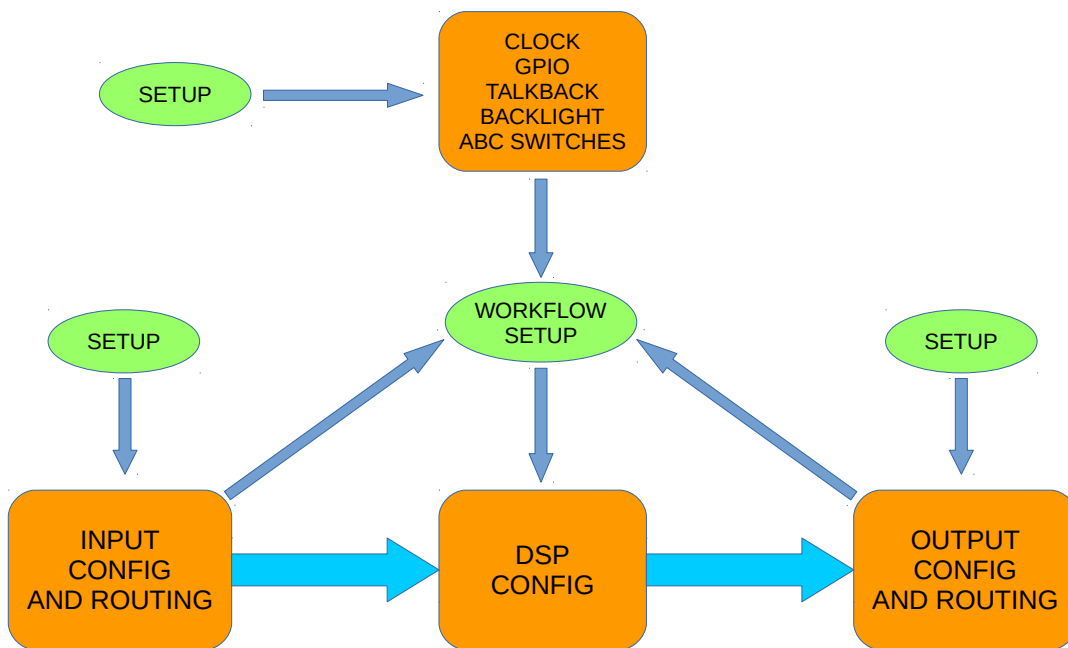

Illustration 34: Workflow Setup Flowchart

The m908 contains a large set of workflow templates for the most common speaker system formats and typical input and output configurations. Selecting one of these templates as a starting point for your system setup can save much time and effort in manual configuration.

System Workflows contain ALL of the configuration settings in the m908. System Workflows can be saved to a USB storage device for making a backup of the system settings. This also

makes it easy to load a specific setup in to multiple m908s.

6.2 Workflow Setup

The m908 ships from the factory with one workflow installed. This workflow is called 7.1.4. If this matches your speaker system requirements then you can skip Workflow Setup and continue to the System Setup section of this manual. If you want to load a new Workflow from a factory template or modify the existing workflow then continue with Workflow Setup.

NOTE: Changes made to the current workflow are not updated in real time. If changes are made they will not become active until the workflow is saved.

⚠️   IMPORTANT: There are several Workflow Setup operations where the ACU will save workflow files in the background. Do not turn the m908 power off while in Workflow Setup or the workflow file can be corrupted.

6.2.1   Workflow Setup Main Screen

To setup the system workflow push long-hold the SETUP switch (for 3 seconds). The SETUP switch will start flashing quickly and the display will have red colored bars across the top and bottom of the screen. When in workflow setup, monitor control functions are held in their current state and cannot be changed until workflow setup is exited.

The top bar displays the message: CHANGES WILL RECONFIGURE SYSTEM
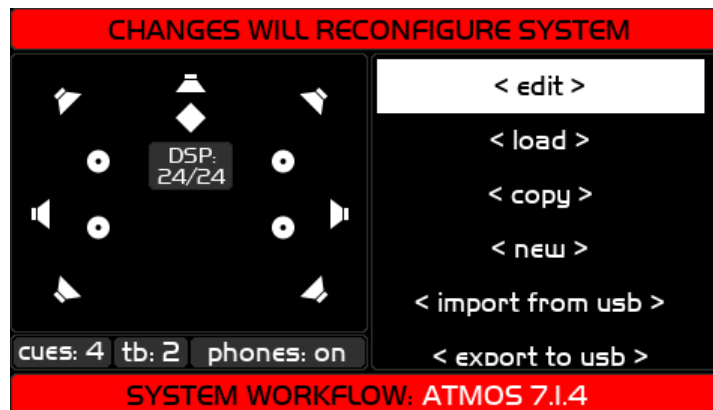
The bottom bar displays: SYSTEM WORKFLOW: *<CURRENT WORKFLOW>*

Illustration 35: Main Workflow Setup Screen

The Workflow Setup screen is divided in to two main windows. In the left window is a graphic representation of speaker system. At the bottom, there are indicators showing the number of stereo cue channels, talkback mics, and headphone system status. In the middle of this window is an indicator displaying the DSP channel utilization for the active workflow.

The right window contains a menu list with the following options:

1. <edit> Edit a workflow
2. <load> Load an existing workflow
3. <copy> Copy an existing workflow to a new workflow
4. <new> Create a new workflow from a list of factory workflow templates
5. <import from usb> Import a workflow from a file saved on a USB storage device
6. <export to usb> Save a workflow to a USB storage device
7. <delete> Delete a workflow

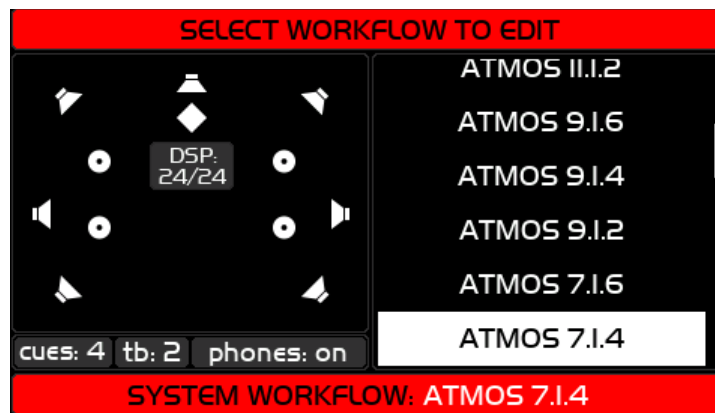Use the VOLUME knob to scroll up and down the list. Push the VOLUME knob to select a menu item.

6.2.2   Select Workflow To Edit Screen

Illustration 36: Select Workflow to Edit Screen

Selecting <edit> from the main workflow setup screen enters this screen.

The Select Workflow To Edit Screen screen is divided in to two main windows.

In the right hand window there is a list of all of the available user workflows. If this is a new m908 there will be only one workflow listed (ATMOS 7.1.4) The currently loaded workflow will be highlighted in the list. At the top of the list is the <exit> menu option.

In the left window is a graphic representation of all of the assigned speaker locations for the workflow that is highlighted in the list to the right. At the bottom are three small windows that indicate the following status for the currently highlighted workflow:

1. cues:0-8
2. tb:0,2
3. phones:on-off

To edit a workflow: scroll to the desired workflow and push the VOLUME encoder. This will open the selected workflow for editing.

To exit the workflow edit window: scroll to <exit> and push the VOLUME encoder.
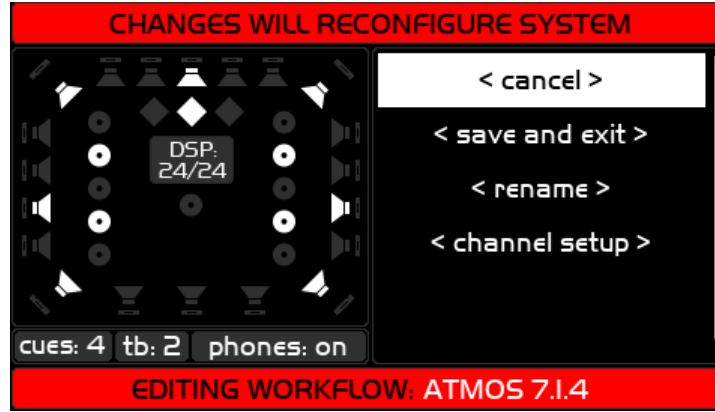
6.2.3 Editing Workflow Screen

Illustration 37: Editing Workflow Screen

In the right window the following menu items are:
1. <cancel> Returns to the previous screen. Asks if you want to exit without saving.
2. <save and exit> Saves the changes to the edited workflow and returns to the previous screen. Asks if you want to save your changes.
3. <rename> Opens the workflow name editing screen with keyboard.
4. <channel setup> Enters channel setup mode where speakers, cues, talkback mics and headphones can be added or subtracted from the workflow.

In the left window **all** of the available speaker positions will be visible. If a speaker position is enabled for the current workflow it will be white. If it is not enabled it will be gray. At the bottom of this window the number stereo cues, talkback mics, and headphone status is displayed.

Speaker Names		
FRONTS	**HEIGHT FRONT**	**TOPS**
LEFT	LEFT HT	LEFT FRONT 1 TOP
LEFT CENTER	LEFT SCREEN HT	RIGHT FRONT 1 TOP
LEFT SCREEN	LEFT CENTER HT	LEFT FRONT 2 TOP
CENTER	CENTER HT	RIGHT FRONT 2 TOP
RIGHT SCREEN	RIGHT CENTER HT	LEFT MIDDLE TOP
RIGHT CENTER	RIGHT SCREEN HT	CENTER MIDDLE TOP
RIGHT	RIGHT HT	RIGHT MIDDLE TOP
		LEFT BACK 1 TOP
SIDE SURROUNDS	**HEIGHT SURROUND**	RIGHT BACK 1 TOP
LEFT WIDE	LEFT WIDE HT SURR	LEFT BACK 2 TOP
RIGHT WIDE	RIGHT WIDE HT SURR	RIGHT BACK 2 TOP
LEFT FRONT SURR	LEFT FRONT HT SURR	
RIGHT FRONT SURR	RIGHT FRONT HT SURR	**SUBS**
LEFT MIDDLE SURR	LEFT MID HT SURR	LEFT SUB
RIGHT MIDDLE SURR	RIGHT MID HT SURR	CENTER SUB
LEFT BACK SURR	LEFT BACK HT SURR	RIGHT SUB
RIGHT BACK SURR	RIGHT BACK HT SURR	
BACKS	**HEIGHT BACK**	
LEFT BACK	LEFT HT BACK	
LEFT CENTER BACK	LEFT CENTER HT BACK	
CENTER BACK	CENTER HT BACK	
RIGHT CENTER BACK	RIGHT CENTER HT BACK	
RIGHT BACK	RIGHT HT BACK	

6.2.4   Workflow Channel Setup Screen

Scroll to the <channel setup> menu option and push the VOLUME encoder to enter the channel setup screen.
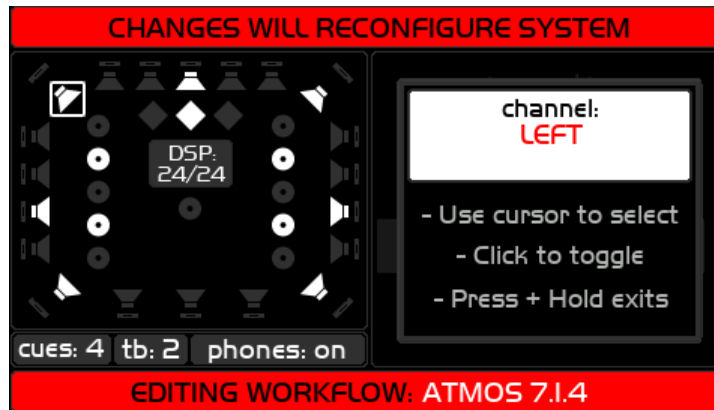
Illustration 38: Workflow Channel Setup Screen

In the channel setup screen the VOLUME encoder controls a highlight cursor that can be moved from speaker to speaker through all of the possible speaker positions. When a speaker is highlighted push the VOLUME encoder to enable/disable that speaker position. When the speaker is enabled it will be white. When the speaker is disabled it will be gray. The name of the currently selected speaker will be shown in the right hand window.

⚠ It is highly recommended that channels L, C, and R always be enabled in any workflow. Failure to have these three speakers enabled can lead to unpredictable behavior of the MONO/L-R functions.

The small window in the middle of the speaker icons shows the current tally of DSP channel utilization. It will update with the number of DSP channels allocated as you add or subtract speakers. Continue scrolling through all of the speakers to get to the cue, talkback, and headphones setup. When one of these switches is highlighted pushing the VOLUME encoder will toggle through the following options:

- **cues:** Each push of the encoder increases the number of stereo cue paths to a maximum of 8 stereo cues. The number of cue channels is limited by the available DSP channels. For example, if there are only 4 DSP channels available then pushing the VOLUME encoder will toggle through cues:0, cues:1, and cues:2.
- **tb:** Each push of the VOLUME encoder toggles the talkback function on or off. The talkback system require 2 DSP channels.
- **phones:** Each push of the VOLUME encoder toggles the headphone system on or off. The headphone outputs require 2 available DSP channels. If phones are enabled the DAC output channels that feed the CUE/EXT connector channels 7-8 are routed to the headphone amplifiers. These channels will not be available for routing to Control Room

speaker systems but will remain active and output the headphone signal on CUE/EXT. channel 7-8. See the m908 System Block Diagram for a picture of this routing.

To exit the channel setup screen, push-hold the VOLUME encoder.

6.2.5   Editing Workflow Rename Screen

From the Editing Workflow screen clicking the <rename> menu option will open the Workflow Rename Screen. This uses Standard Name Editing interface. If a name is chosen that is already used by another existing workflow the system will not overwrite the existing workflow. A dialog box will appear indicating the conflict.
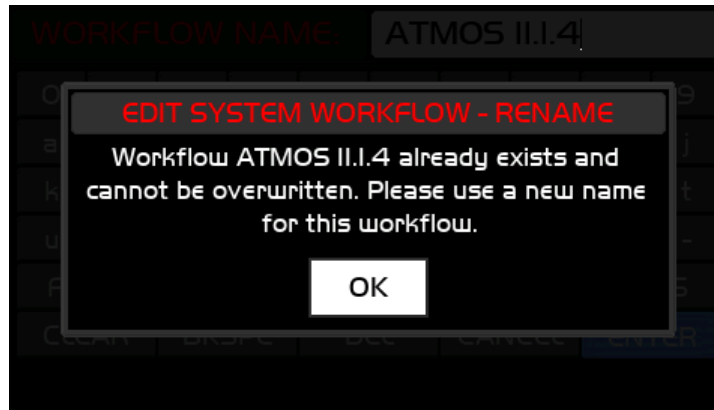
Illustration 39: Editing Workflow Rename Conflict

Clicking OK in this dialog will return to the Workflow name editing screen so that another name can be entered.

To cancel Workflow Rename select the CANCEL button on the rename keyboard.

6.2.6   Editing Workflow Save and Exit

From the Editing Workflow screen clicking the <save and exit> menu option will open a dialog box asking if you are sure you want to save changes made to the workflow open for editing. This dialog appears whether or not any changes were made.
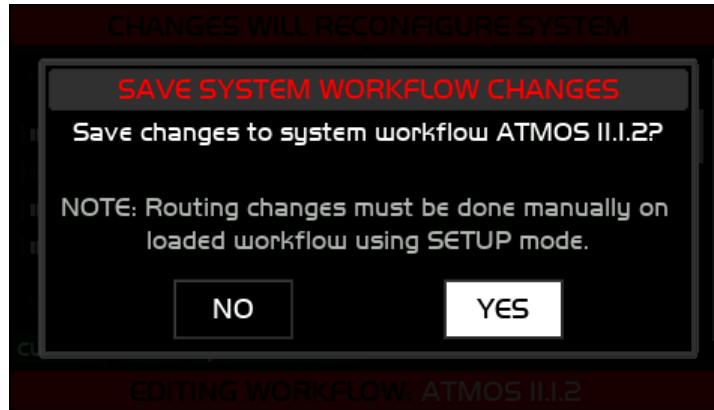
Illustration 40: Save System Workflow Changes Dialog

⚠ If the workflow being edited is the currently active system workflow the changes will be executed upon exiting. All monitor levels will be set to zero.
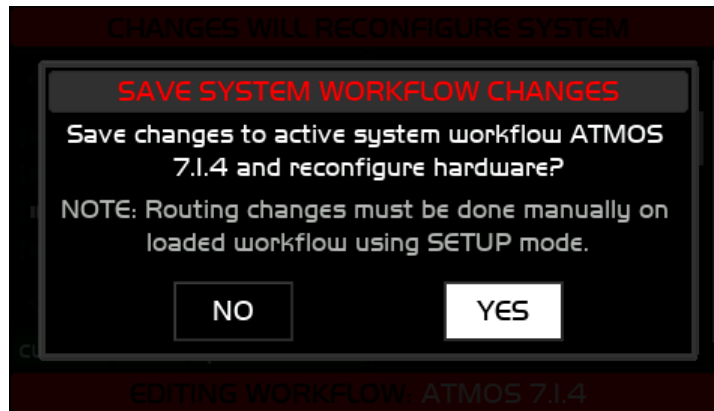
Illustration 41: Save Active System Workflow Changes Dialog

Clicking YES will execute the changes. While the System Workflow is being updated and reloaded the following message will appear:
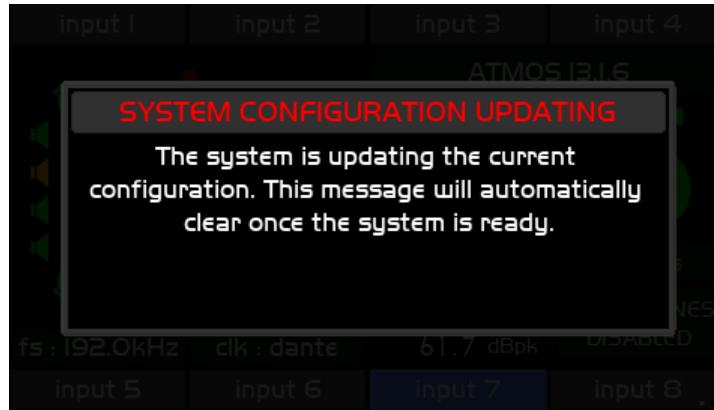
Illustration 42: System Configuration Updating

After the system is updated the *editing workflow* screen will appear.

Exiting from the *editing workflow* screen returns to the *main workflow setup* screen. To exit this screen and return to the *home* screen press and release the flashing SETUP switch.

Note that if any changes have been made to the workflow that affect the number of speakers, their positions, number of cue channels, talkback mics, or headphone status then there will need to be routing changes made in the regular setup menus to reconnect hardware inputs and outputs to the changed DSP channels.

If changes were made to a workflow that is not the currently loaded system workflow then that workflow would need to be loaded so that routing changes can be made.

Changes to the workflow name will not have any effect on routing.

6.2.7   Editing Workflow Cancel

From the Editing Workflow screen clicking the <cancel> menu option will open a dialog box asking if you are sure you want to discard changes.
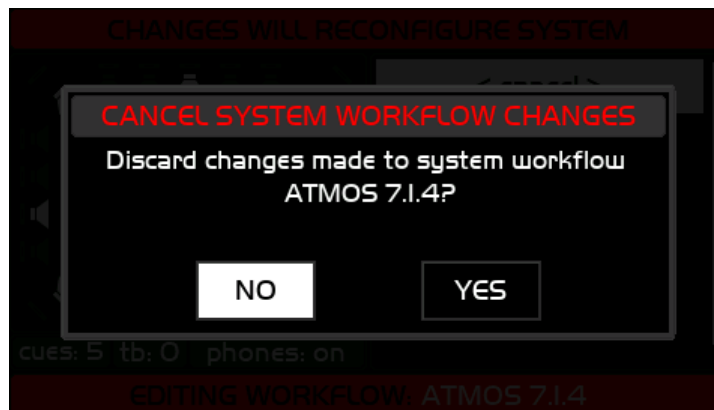
Illustration 43: Cancel System Workflow Changes Dialog

Selecting "NO" will return to the workflow editing screen.

Selecting "YES" will exit to the *main workflow setup* screen.

To return to the *home* screen press-release the flashing SETUP switch.
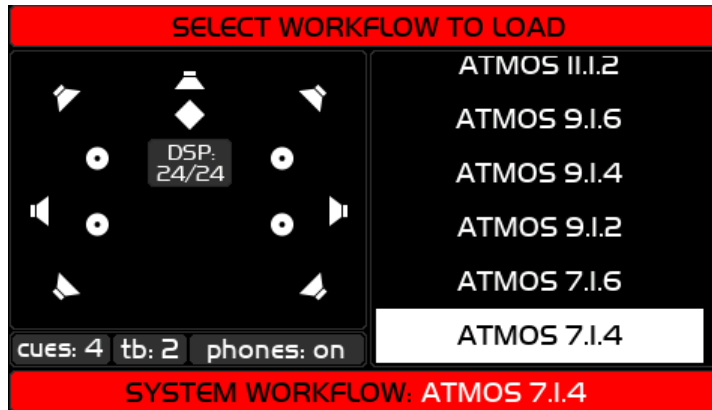
6.2.8   Workflow Load

Illustration 44: Selected Workflow to Load Screen

The Select Workflow To Load screen displays a list of the Workflow presets that have been saved on the m908. While scrolling through the list of workflows the DSP channel allocations will be displayed on the left window. Push the VOLUME encoder to select a workflow. A confirmation dialog box will appear asking if you want to load the selected preset.
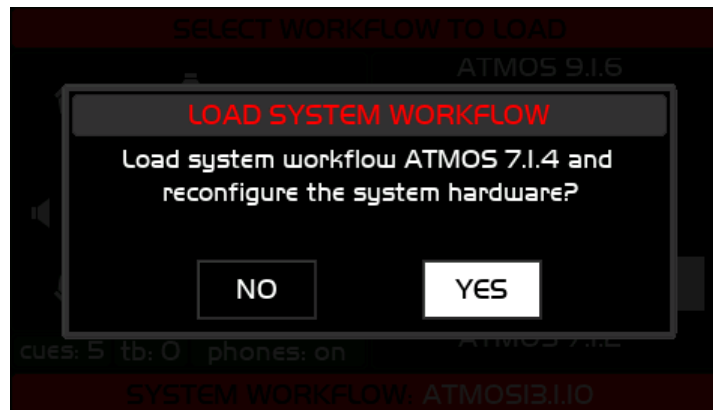
Illustration 45: Load System Workflow Dialog

Select YES to load the workflow in to the m908. The system will be updated. Select NO to go back to the Select Workflow To Load screen.

⚠ If YES is selected the system will be reconfigured to the new workflow and all monitor levels will be set to zero.

Select <exit> to exit the Select Workflow To Load screen without loading a new workflow.
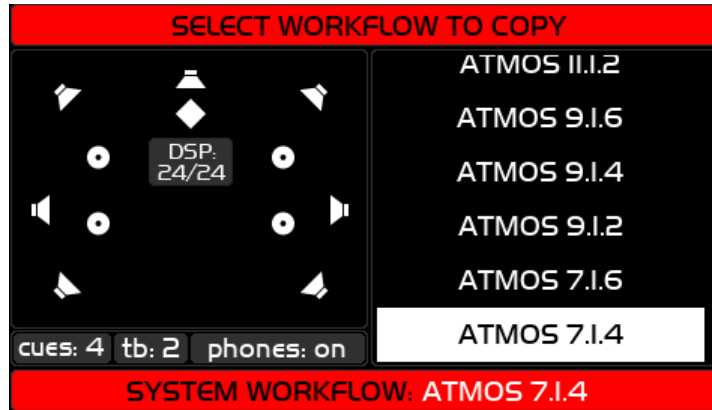
6.2.9   Workflow Copy

Illustration 46: Select Workflow to Copy Screen

The Select Workflow To Copy screen displays a list of the Workflow presets that have been saved on the m908. While scrolling through the list of workflows the DSP channel allocations will be displayed on the left window. Push the VOLUME encoder to select a workflow. A confirmation dialog box will appear asking if you want to copy the selected preset.

Select YES to copy the workflow. The list of workflows will now include the new copy.

Select NO to go back to the Select Workflow To Copy screen.

The new copy will have the name of the original with -1 added to the end. For example, copying the workflow "ATMOS 7.1.4" will make the workflow "ATMOS 7.1.4-1". If "ATMOS 7.1.4-1" already exists then the new file will be named "ATMOS 7.1.4-2" and so on.

Select <exit> to exit the Select Workflow To Load screen without loading a new workflow.
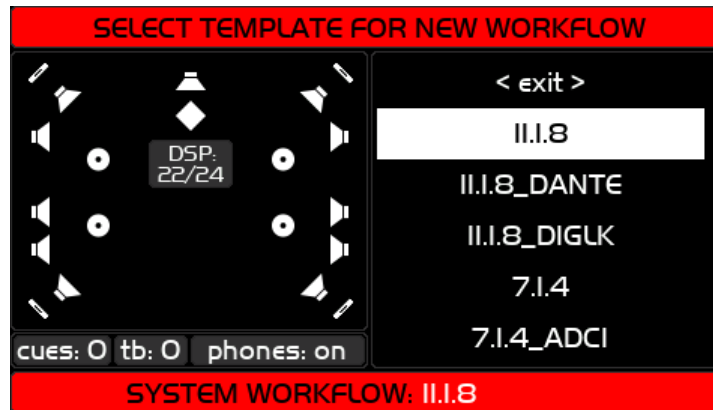
6.2.10   Workflow New

Illustration 47: Select Template for new Workflow

The Select Template For New Workflow screen displays a list of the factory workflow templates. These are read-only and can be loaded in to the systems list of available workflows. They can then be modified, renamed, and saved.

These templates are based on the most common surround speaker system formats and contain default input and output configurations and signal routing.

While scrolling through the list of workflow templates the DSP channel allocations will be displayed on the left window. Push the VOLUME encoder to select a workflow template. Selecting a workflow template will open a confirmations dialog box asking if you want to create a new system workflow.
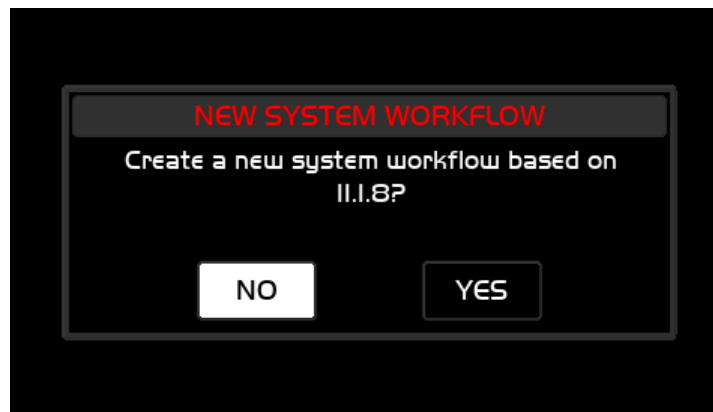
Illustration 48: New System Workflow Confirmation

Click NO to return to the Select Template for New Workflow screen.

Click YES to create new workflow. This will add the new workflow to the list of available user workflows. You can load this workflow in to the m908 by using the <load> option.

The new workflow will have the name of the template. If a workflow already exists with that

name then "_1" added to the end. For example, creating the workflow "ATMOS 7.1.4" will make the workflow "ATMOS 7.1.4". If "ATMOS 7.1.4" already exists then the new file will be named "ATMOS 7.1.4_1".

Select <exit> to exit the Select Template For New Workflow screen without loading a new workflow.

The m908 comes with a variety of workflow templates designed to help speed up configuration and installation. The templates are designed to work with various m908 hardware configurations and come with predefined Input and CR Output routing. The table below lists the workflow template naming convention for various m908 hardware configurations.

Workflow Templates	
Workflow Name	m908 Option Configuration
x.x.x	Base
x.x.x_ADC1	ADC1
x.x.x_ADC2	ADC1, ADC2
x.x.x_ADC1DA	ADC1, DANTE
x.x.x_ADC1-DL	ADC1, Digilink
x.x.x_ADC2DA	ADC1, ADC2, DANTE
x.x.x_ADC2-DL	ADC1, ADC2, Digilink
x.x.x_DANTE	DANTE
x.x.x_DIGILK	Digilink

The specific system wide settings for each workflow template are available in spreadsheet format. This makes it easier to identify all of the Input and Output routing and system settings built in to each of the workflow templates. The spreadsheets can be downloaded from the product manual tab of Support section of the Grace Design web site.

https://gracedesign.com/support-documents/

6.2.11   Workflow Import From USB

The m908 can export and import workflows using a USB drive plugged in to the USB A connector on the rear panel of the RCU. The USB device must be formatted FAT16 or FAT32 for the RCU to recognize it. Workflows are saved in a json formatted text file with a .gdw file name extension.

Being able to export and import workflows makes it possible create backups of a system setup as well as making it easy to copy system setups to multiple m908s.
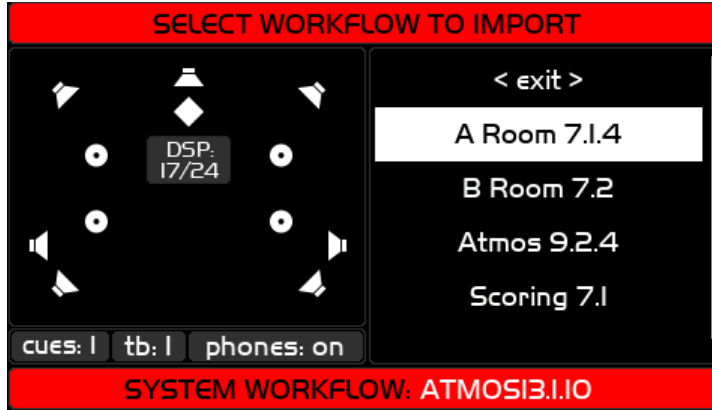

Illustration 49: Select Workflow to Import from USB

The Select Workflow To Import screen displays a list of workflow files located on the attached USB storage device.

While scrolling through the list of workflows on the USB device DSP channel allocations will be displayed on the left window. Push the VOLUME encoder to select a workflow to import. Selecting a workflow will open a confirmations dialog box asking if you want to import the selected workflow.
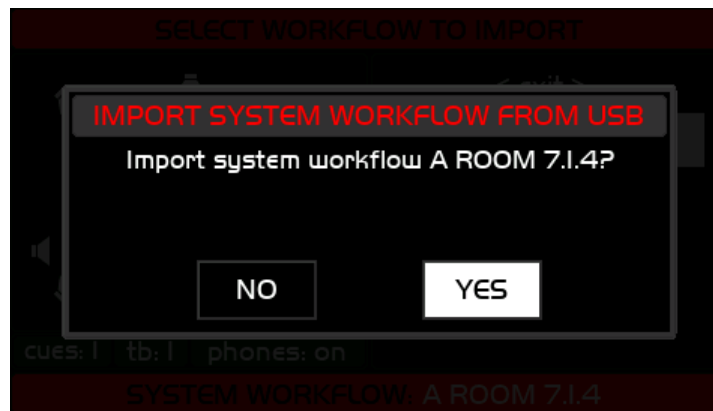
Illustration 50: Import Workflow from USB Dialog

Select YES to import the workflow from the USB device to the m908. The following message will appear while the file is being transferred.
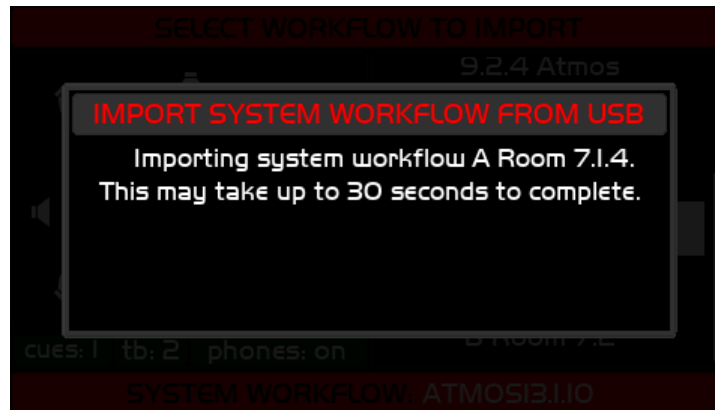
Illustration 51: Import Workflow from USB Status

The list of workflows will now include the imported workflow.

If the name of the workflow being imported is the same as a workflow on the m908 already then the imported workflow will have "_1" appended to the name. If there is already a workflow with "_1" appended to the name then the newly imported workflow will be named with "_2" appended to the name.

Select NO to go back to the Select Workflow To Import screen.

Select <exit> to exit the Select Workflow To Import screen without importing a new workflow.

In the unlikely event that the m908 experiences a USB file system error the following dialog will appear.
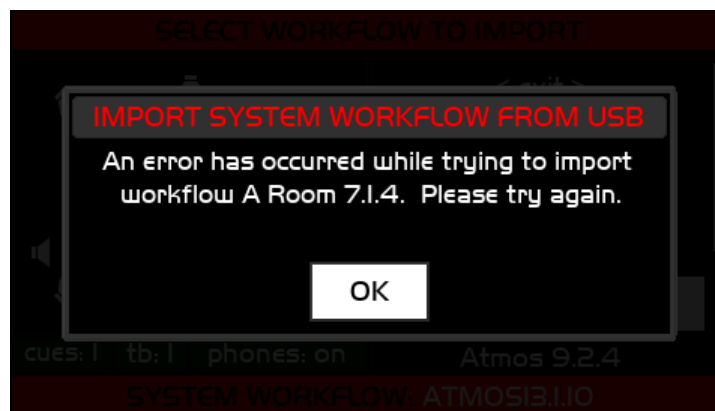
Illustration 52: Import Workflow from USB Error Dialog

If you encounter this error message click OK and try again. If the problem persists contact Grace Design Technical Support.

6.2.12   Workflow export to USB

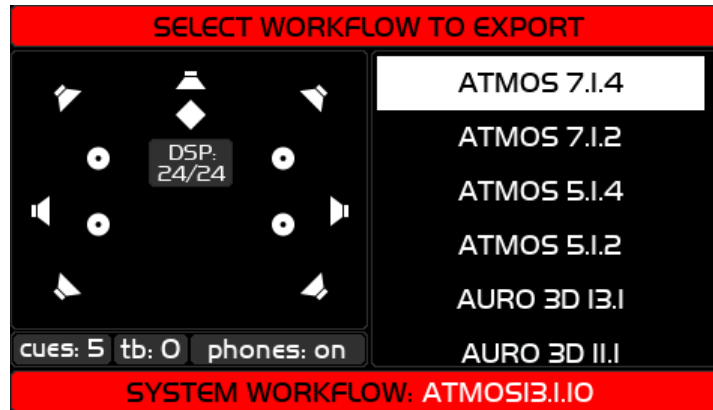The Select Workflow To Export screen displays a list of workflow files located on the m908 available for export.

Illustration 53: Export Workflow to USB

⚠️ It is highly recommended to make backups of your workflows on a USB drive for safe keeping. In the unlikely event that a workflow file gets corrupted to power loss you will be able to quickly restore your system settings from the backup.

⚠️ Do not rename an exported workflow file on your computer. Doing so will make the workflow file unreadable by the m908. Only rename workflows using the m908 Workflow Rename function.

While scrolling through the list of workflows DSP channel allocations will be displayed on the left window. Push the VOLUME encoder to select a workflow to export. Selecting a workflow will open a confirmations dialog box asking if you want to export the selected workflow.
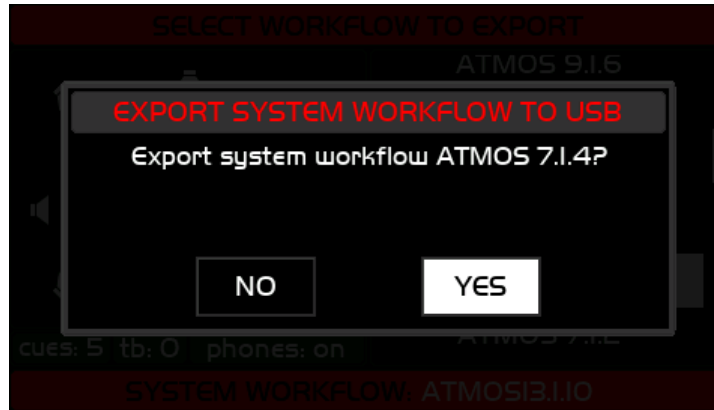
Illustration 54: Export Workflow to USB Confirmation

Select YES to export the workflow from the m908 to the USB storage device. The following message will appear:
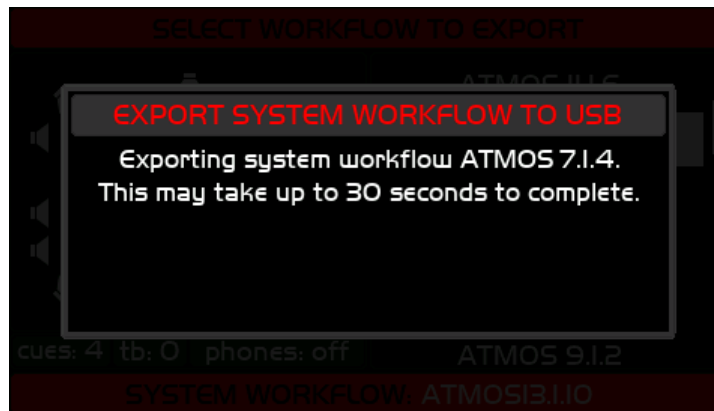

Illustration 55: Export Workflow to USB Status

Select NO to go back to the Select Workflow To Import screen.

Select <exit> to exit the Select Workflow To Import screen without importing a new workflow.

If there is no valid USB memory device plugged in the the RCU the system will display the following message when an export to USB is attempted.

Illustration 56: Insert Proper USB Drive Message

Make sure that the USB memory device is inserted and that it is formatted FAT16 or FAT32.

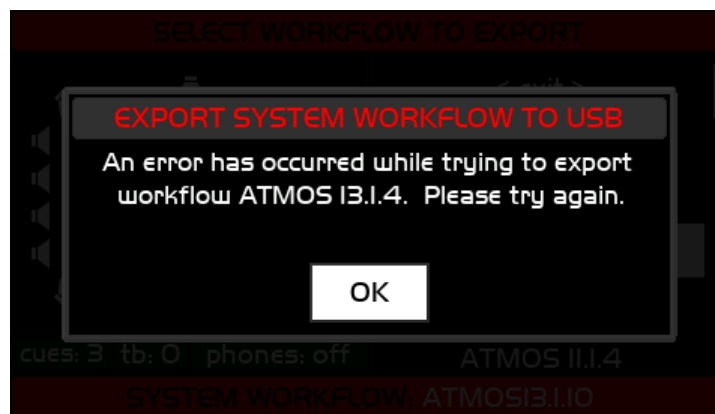In the unlikely event that the m908 experiences a USB file system error the following dialog will appear.

Illustration 57: Export Workflow to USB Error

If you encounter this error message click OK and try again. If the problem persists contact Grace Design Technical Support.

6.2.13   Workflow delete

The Select Workflow to Delete screen list the available workflows on the system. Scroll to the workflow to delete and click.
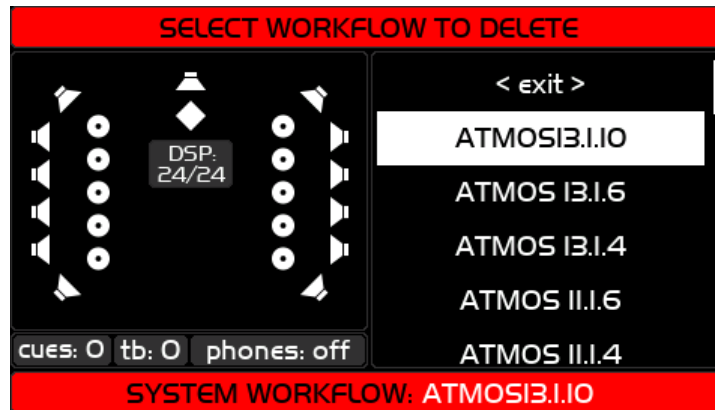
Illustration 58: Select Workflow to Delete

A confirmation dialog will appear. Click NO to cancel deleting the selected workflow. Click YES to permanently delete the selected workflow.
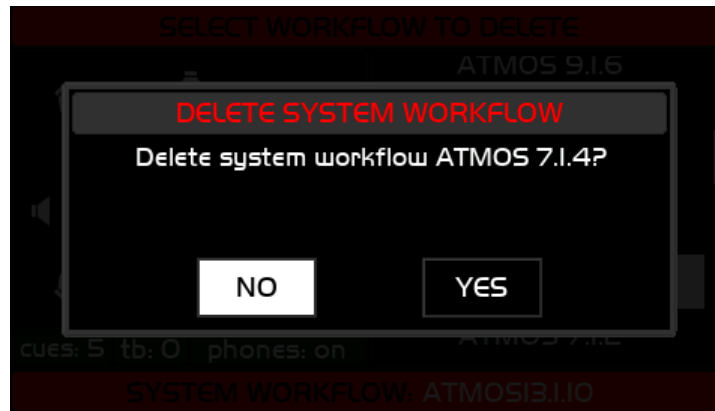

Illustration 59: Delete Workflow Confirmation

Note: The *currently* loaded and active workflow can not be deleted. To delete the active workflow first load a different workflow. If you attempt to delete the current workflow this message will appear:
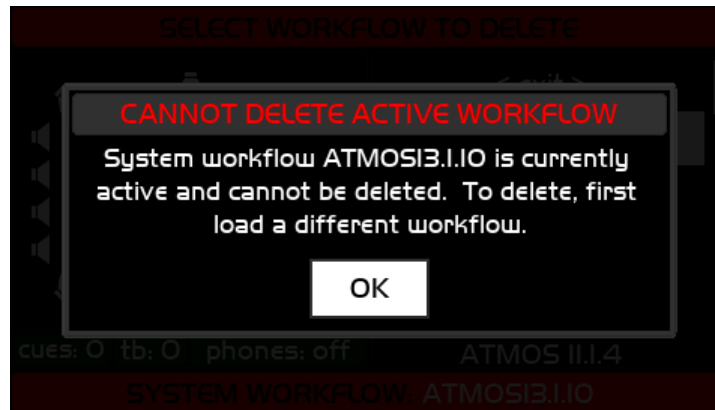
Illustration 60: Cannot Delete Active Workflow

6.3 Setup

6.3.1 Navigating and Using the Setup Menus

From the home screen pushing the SETUP switch once enters the m908 in to Setup Mode. The SETUP switch will flash slowly and the following screen will be displayed:
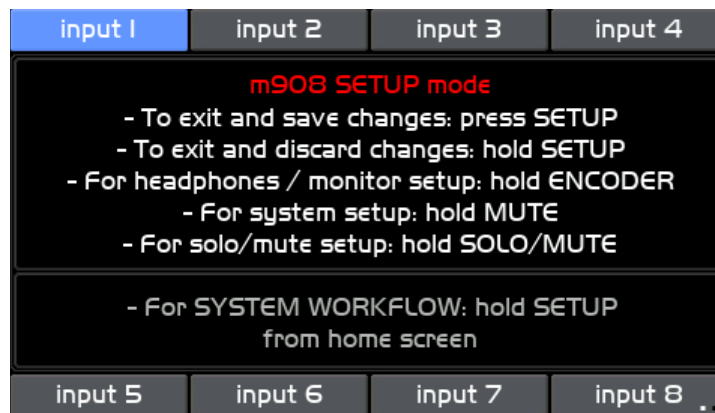

Illustration 61: Initial Setup Mode Dialog

A single push of the SETUP switch will exit Setup Mode and save any changes that were made.

⚠️ IMPORTANT: After exiting Setup Mode the ACU will save the workflow file in the background. This can take up to 5 seconds. Do not turn the m908 power off during this time or the workflow file can be corrupted.

A push long-hold of the SETUP switch will exit Setup Mode and prompt you to discard any changes that were made. Clicking NO will return to the previous Setup screen. Clicking YES will
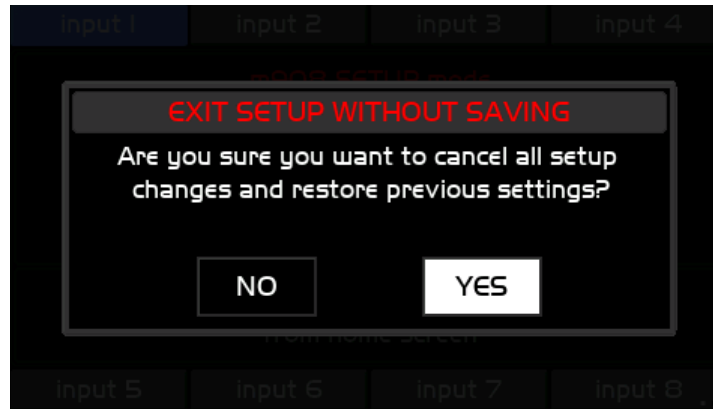
exit Setup without saving changes.

Illustration 62: Exit Setup Without Saving Dialog

Pushing any of the following switches will enter in to the setup for their specific function:
- INPUT SELECT 1-8
- DIM
- MONO
- MON>CUE
- MUTE
- CR1-3
- TALKBACK
- A, B, and C User Switches

  Press-hold any of the following switches will enter in to the setup for their specific function:
- VOLUME for Monitor
- MONO for Left minus Right
- MON > CUE for SPL
- MUTE for System
- SOLO/MUTE for Solo / Mute

When in the setup screen for any of these functions the parameters can be changed and auditioned in real time.

At any time another switch can be pushed to switch to another setup screen.

Save any changes by pushing the SETUP switch and exiting Setup Mode.

The following section outlines the available Setup parameters in the m908.

6.3.2   Standard Parameter Editing

Throughout the m908 Setup menus there are lists of parameters that can be adjusted/edited. In each menu, the current mode is displayed in red text in the upper left. The conventional behavior is that turning the VOLUME encoder will scroll through this list while highlighting the available parameters. The parameter name is on the left and the value is on the right. For instance an input level offset will look like this:
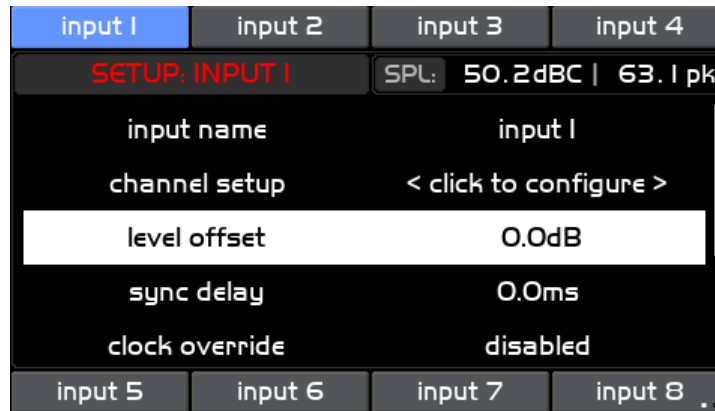

Illustration 63: Edit Screen With Parameter Highlighted but Not Selected for Editing

Pushing the VOLUME encoder will enable the parameter for editing and the value will turn red. Turning the VOLUME encoder will change the value of the parameter.
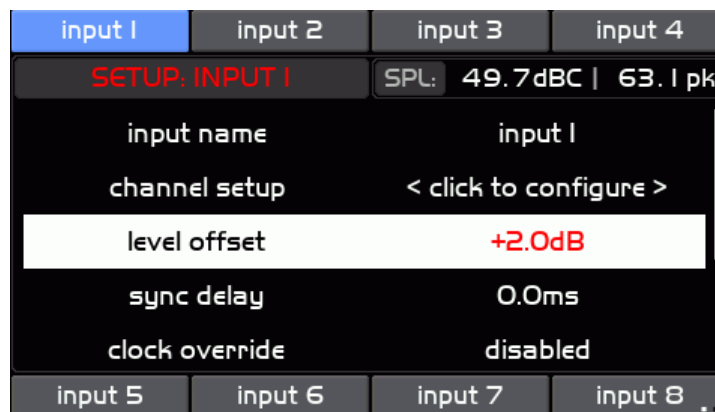

Illustration 64: Edit Screen with Parameter Highlighted and Selected for Editing

Pushing the VOLUME encoder again will exit editing of the parameter and the value will turn black. From here, turning the VOLUME encoder will again scroll through the list of parameters.

6.3.3   Standard Name Editing

m908 features a name entry and editing keyboard interface. It is used in any setup menu where a name can be created or customized. Some examples are:

- INPUT Name
- SPEAKER Name
- WORKFLOW Name
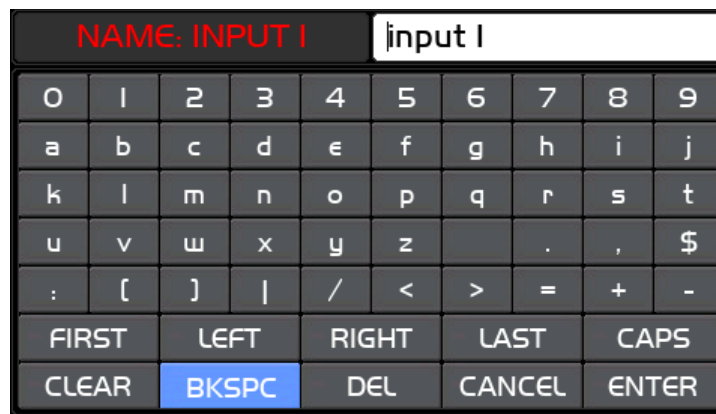- USER SWITCH Name
- System PASSWORD

Illustration 65: Name Edit Screen

For features that only require numbers there is a simplified Numeric Input Screen. It is used for entering Ethernet IP address and subnet mask numbers.
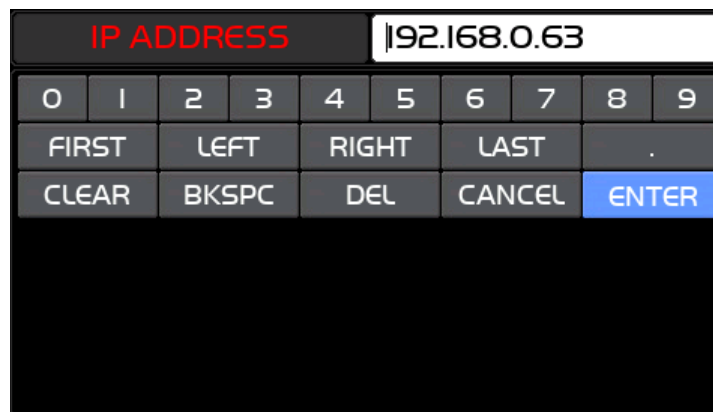

Illustration 66: Number Edit Screen

In the *both* screens the current name is displayed in the text edit window in the upper right portion of the screen. The maximum number of characters for a custom name is 12.

The upper section of the keyboard contains numbers, letters, and special characters. The lower section of the keyboard contains the following text editing tools:

1. FIRST   Moves the cursor to the left most position in the text edit window
2. LEFT   Moves the cursor one character to the left
3. RIGHT Moves the cursor one character to the right
4. LAST   Moves the cursor to the right most position of the current test string
5. CAPS   Toggles between upper case and lower case letters
6. CLEAR Clears all text from the text edit window
7. BKSPC Removes the character to the left of the cursor
8. DEL   Removes the character to the right of the cursor
9. CANCEL   Exits the *Name* screen without saving changes
10. ENTER   Saves changes and exits the *Name* screen

Note: Pressing the SETUP switch while in the Name/Number edit screen will exit without saving changes. Be sure to select ENTER and click the encoder or push the SOLO/MUTE switch to save the changes and exit.

Navigation:

Turning the VOLUME encoder will scroll through all the keys in the keyboard. The selected key will be highlighted in blue. Pushing the VOLUME encoder will press the highlighted key. The keyboard can also be navigated with the illuminated switches in the SOLO/MUTE section of the RCU front panel. The SUB, LB, RB, and B switches act as up down left right navigation keys. The SOLO/MUTE switch has the same "ENTER" function as pushing the VOLUME encoder.

These switches will be flashing green.
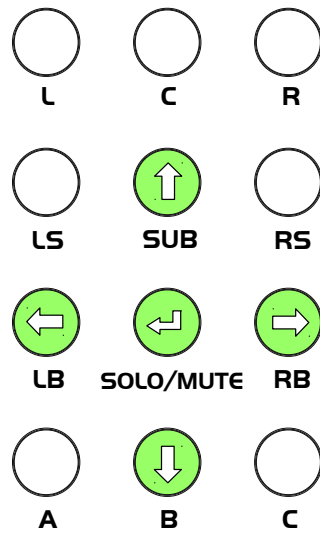
Illustration 67:
Name Edit Navigation Keys

6.3.4   Standard Channel Routing

Several Setup screens contain channel routing options. The following signal types contain routing configurations:

- Input Sources
- Control Room Speakers
- Cue inputs and outputs
- Meter Outputs

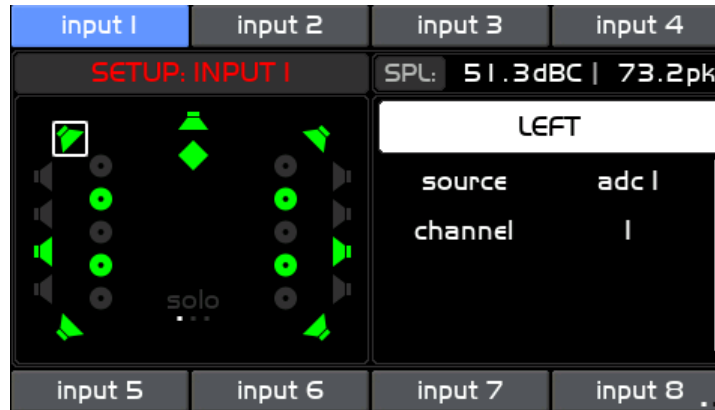The following is an *Input* Channel Routing setup:

Illustration 68: Input Channel Setup Screen

Clicking the channel name enables the channel cursor in the left window. Rotate the encoder to move the cursor to the desired input channel type. The **source** parameter shows the connector being used for the currently selected input. Clicking and scrolling through the list will display the available connectors.

Clicking on a connector that is different from the current connector will disables the current channel selection and immediately mute the audio. Scroll to the **channel** parameter and select the desired channel number from the list. Upon clicking on a channel number the audio will be routed from the new selection.

NOTE: Input connectors can be used for multiple inputs. For instance, you can route *analog 1* channels 1 and 2 to Cue 1 and also route the same channels to an Input. This way you can easily select Cue 1 as a monitor source.

The following **Output** Channel Routing behaviors are the same for Control Room Speaker outputs, and Meter outputs:
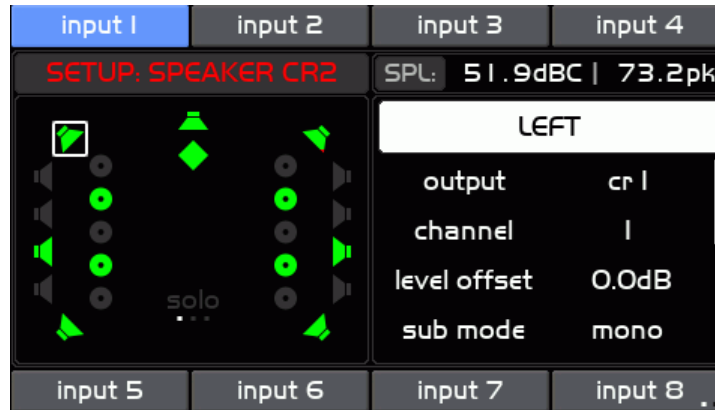
Illustration 69: Speaker Channel Setup Screen

Clicking the channel name enables the channel cursor in the left window. Rotate the encoder to move the cursor to the desired input channel type. The **output** parameter shows the connector being used for the currently selected speaker (signal source). Clicking and scrolling through the list will display the available connectors.

Clicking on a connector that is different from the current connector will disables the current channel selection and immediately mute the audio. Scroll to the **channel** parameter and select the desired channel number from the list. Upon clicking on a channel number the audio will be routed to the new selection.

NOTE: If a connector/channel is selected that in already in use by a Cue, or Meter output then the following dialog will appear:
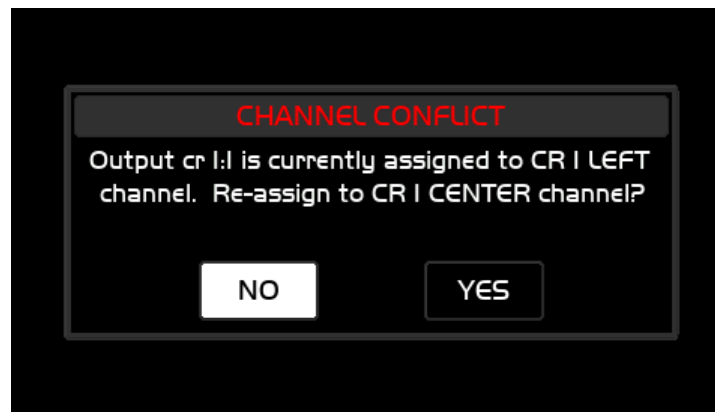

Illustration 70: Channel Routing Conflict Dialog

6.3.5   Input Setup

Push the SETUP switch then any of the 8 INPUT select switches will enter the Setup Input screen.

The m908 has 16 inputs available with 8 inputs visible at a time. To change "input pages" push-hold the lower right INPUT (◄ ►) select switch.
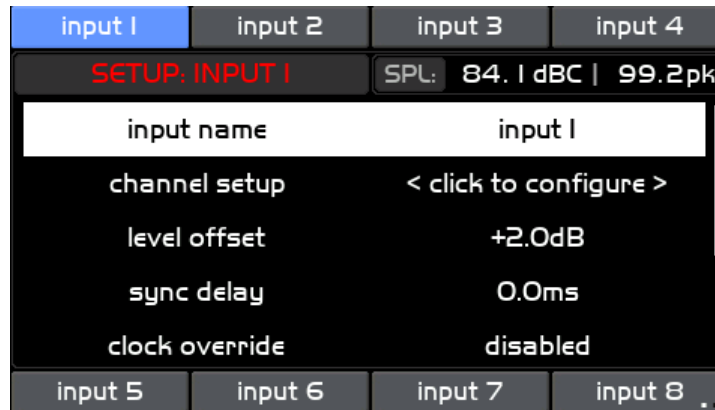

Illustration 71: Setup Input Screen

This screen contains the following list of setup parameters that can be selected by turning the VOLUME encoder knob. The available parameters are:

- **input name** Clicking this option enters the Name Edit Screen. (see above for Standard Name Editing).
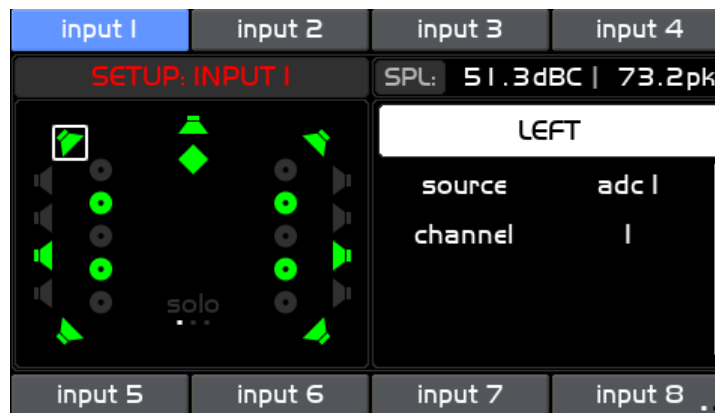- **channel setup** Clicking this option enters the Channel Setup screen.

Illustration 72: Input Channel Setup Screen

The Channel Setup screen is where input channels are enabled and routed to DSP input channels.

The left side of the screen window shows all of the available DSP input channels enabled in the current Workflow. Speaker Icons that are white, green, amber, or red are enabled for the currently selected input. Channels that are disabled are indicated with the gray speaker icons. The currently selected DSP channel has a white box around it.

NOTE: The speaker icons will illuminate green, amber, or red for channels that are used the currently selected CR Output. If an input channel is not used in the currently selected CR Output it will remain white even if signal is present on that input channel.

The three possible Sub channel icons seen in the Workflow Setup or Control Room Setup screens will be highlighted as one LFE channel.

Since an input will only have one LFE channel this is where the LFE signal is routed so that it will be properly handled by the DSP. Input signals routed to the LFE channel will run through the LFE low pass filter and 0-10dB gain option. These settings are in Monitor Control setup.

On the right side of this screen there is a list of options and the top item is highlighted initially:

- ○ **channel name** The name of the currently currently selected DSP channel in the left window. Pushing the VOLUME encoder enters the Channel Select mode. Turning the encoder scrolls through the available DSP channels. Pushing the VOLUME encoder selects the highlighted DSP channel for routing. See the Standard Channel Routing section of this manual for details on channel routing.

- ○ **source** Selects the input connector for the currently selected DSP channel. Select "disabled" if the selected DSP channel is not used in the current Input. The selected speaker icon will turn gray.
- ○ **channel** Select the desired connector channel for the currently selected DSP channel. Select "disabled" if the selected DSP channel is not used in the current Input. The selected speaker icon will turn gray.

To exit the Channel Setup screen and return to the Setup Input screen press the INPUT select switch for the currently selected input.

- • **level offset** Push the VOLUME encoder to adjust the offset for the selected input. The range is -20dB to +20dB in 0.5dB steps. Note that Input level offset affects all channels in the input source equally.
- • **sync delay** A delay (lip sync delay) can be added to an input with this parameter. The maximum amount of delay is 1000ms. The delay increment is 5.0ms
- • **clock override** Select the input source as clock source for the system. Options include

<disabled, *connector-name*>

When an input is defined it can be selected to override the current system clock source when it is selected for monitoring. This can be useful if the input source can not be synchronized to an external clock. If the input being defined spans across more than one hardware connector make sure that both input connectors are at the same sample rate and are syncronous. The clock override source can be any clock source except wordclock.

NOTE: Inputs that are defined to have a clock source override can not be included in input summing unless the other inputs have the same clock source override.

•

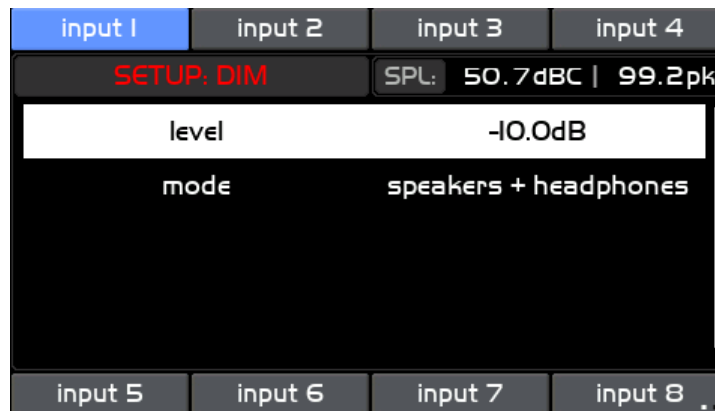Push the SETUP switch to save and exit Setup Mode.

6.3.6   Dim setup

Illustration 73: Setup Dim Screen

Push the SETUP switch then the DIM switch to enter Dim setup. If DIM was active before pushing SETUP it will remain active. If DIM was not active then press the DIM switch to activate the DIM function for auditioning.

This screen contains the following a list of setup parameters that can be highlighted by turning the VOLUME encoder knob. The available parameters are:

• **level** Clicking this option adjusts the amount of level attenuation when the Dim is active. The range is 0dB to -30dB in 0.5dB steps.

• **mode** The mode options select which monitor paths will dim when Dim is active. The options are:

speaker + headphones

speakers only

headphones only

The DIM switch is active in Setup mode so the Dim settings can be auditioned.

Push the SETUP switch to save and exit Setup Mode.

6.3.7   CR1, CR2, CR3 Speaker setup

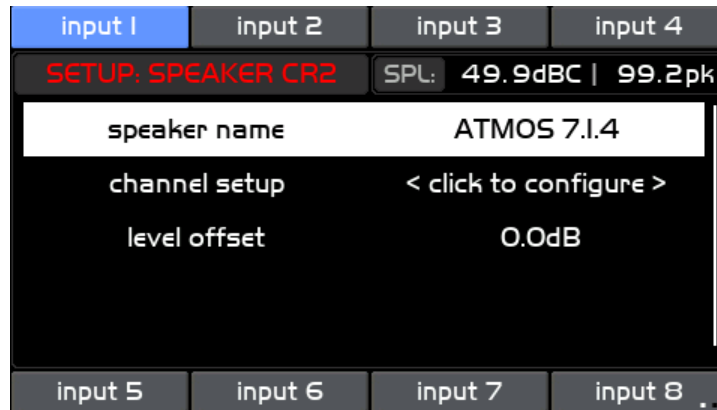Push the SETUP switch and then any of the CR1, CR2, or CR3 speaker select switches to enter the Setup Speaker screen.

Illustration 74: Setup Speaker CR Screen

This screen contains the following list of setup parameters that can be highlighted by turning the VOLUME encoder knob. The available parameters are:

- **speaker name** Clicking this option enters the Name Edit Screen. (see navigating the Standard Name Editing screen).
- **channel setup** Clicking this option enters the Channel Setup screen.
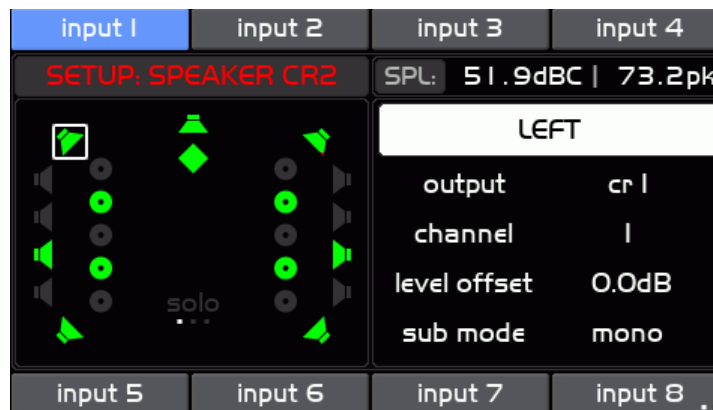

Illustration 75: Speaker Channel Setup

The Channel Setup screen is where speaker output channels are enabled and routed from DSP speaker output channels. As well, delay, level offset, and bass management parameters are adjusted here.

The left side of the screen window shows all of the available speaker locations enabled in the current Workflow. Speaker Icons that are white (or green if signal is present) are enabled for the currently selected Control Room output. Speakers that are disabled are gray. The currently selected speaker channel has a white box around it.

On the right side of this screen there is a list of options and the top item is highlighted initially:

- **channel name** The name of the currently currently selected channel in the left window. Pushing the VOLUME encoder enters the Channel Select mode. Turning the encoder scrolls through the available channels. Pushing the VOLUME encoder selects the highlighted channel for routing etc. See the Standard Channel Routing section of this manual for details on channel routing.

- **output** Selects the speaker output connector for the currently selected speaker. Select "disabled" if the selected speaker is not used in the current speaker system. The selected speaker icon will turn gray.

- **channel** Select the desired connector channel for the currently selected DSP speaker output. Select "disabled" if the selected speaker is not used in the current speaker system. The selected speaker icon will turn gray.

    NOTE: If the selected connector channel is already in use by another speaker channel in the same Control Room speaker system or a Cue or Meter signal a dialog box will appear asking if you want to replace the existing routing with the new selection.
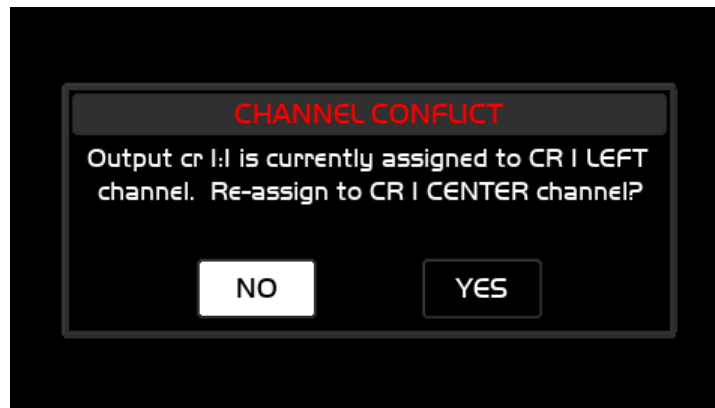

Illustration 76: Channel Routing Conflict Dialog

If the selected connector channel is already in use by another Control Room speaker system this allowed and that physical speaker will be shared between both speaker systems.

Note: CR1, CR2, and CR3 are all relay selected from the same 8 channel DAC. Therefore you can not define a speaker system to contain signals going to the same channel on more than one of these connectors. Channels 1 and 2 are switched together to all three output connectors. Because of this channels 1 and 2 can only be routed to the same connector.
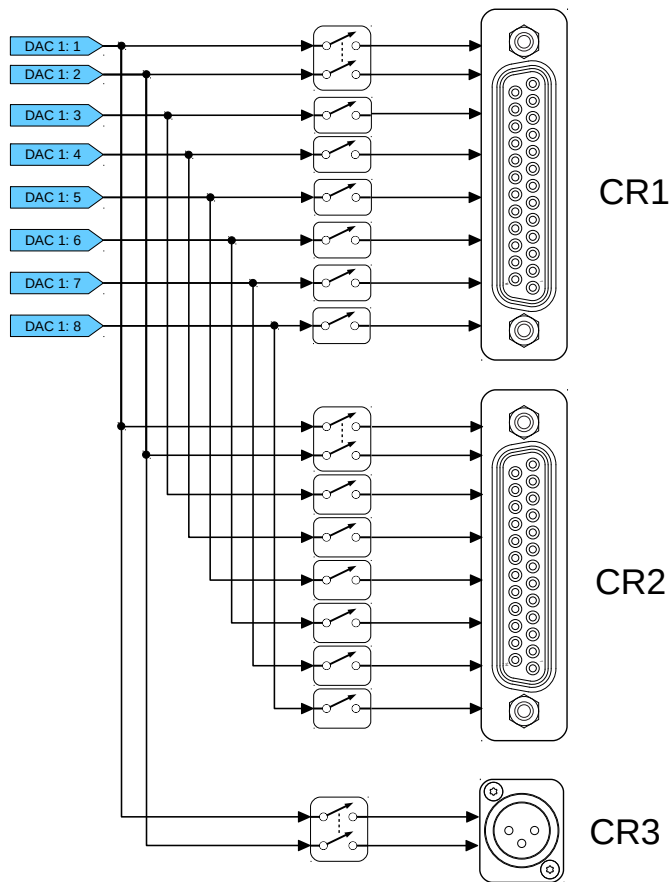

Illustration 77: DAC 1 Output Circuit

Below is a table of connector channels that can not exist in the same speaker system.

Control Room Out Hardware Conflicts		
CR1:1	CR2:1	CR3:1
CR1:2	CR2:2	CR3:2
CR1:3	CR2:3	
CR1:4	CR2:4	
CR1:5	CR2:5	
CR1:6	CR2:6	
CR1:7	CR2:7	
CR1:8	CR2:8	

An attempt to route a speaker to a connector channel that is in conflict with another connector will result in the following type of error messages:
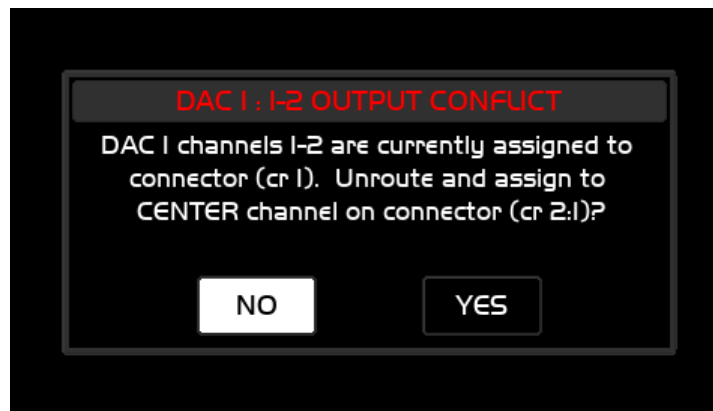

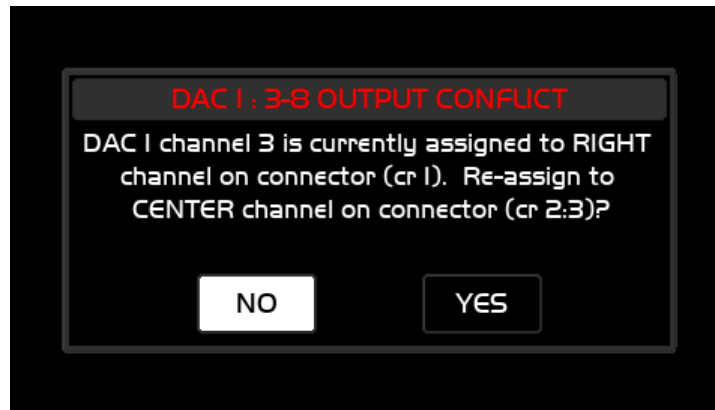*Illustration 78: DAC 1: 1-2 Output Conflict*

Illustration 79: DAC 1: 3-8 Output Conflict

NOTE: If Headphones are enabled in the current Workflow CUE/CR EXT channels 7-8 can not be used for control Room output speakers and will not show up in the list of available outputs. This is because the Headphones use the DAC channels that feed CUE/CR EXT 7-8. If Control Room speakers are routed to CUE/CR EXT channels 7-8 and the workflow is changed to enable headphones previous routes to CUE/CR EXT 7/8 will be disabled.

- **level offset** Volume offset for calibrating level of the selected speaker. The range is <-20.0dB to +20.0dB> in 0.5dB steps.
- **delay** Delay for time alignment calibration can be added to a speaker with this parameter. The maximum amount of delay is 250ms. The delay increment is ~0.1ms
- **crossover, hpf slope, lpf slope,** See the Bass Management setup section of this manual.
- **room correction** See the Room Correction setup section of this manual.
- **sub mode** Sets the signal source for the selected sub. Options are <left, right, mono>

To exit the Channel Setup screen and return to the Setup Speaker screen push the CRx switch that is currently illuminated.

- **level offset** Sets the level offset calibration for the currently selected Control Room speaker system. Range is <-20.0dB to +20.0dB> in 0.5dB steps.

Press the SETUP switch to save and exit Setup Mode.

6.3.8   Bass Management setup

The m908 bass management configuration is contained in the Control Room Setup screen.

Push the SETUP switch and then any of the CR1, CR2, or CR3 speaker select switches to enter the Setup Speaker screen. Click on **channel setup** to enter the Channel Setup screen.

Illustration 80: Bass Management Setup

The following parameters control the Bass Management processing:

- **crossover** Selects the crossover frequency. Range is <50-150Hz> in 1Hz steps.
- **hpf slope** Sets the high pass filter slope. Options are <bypass><6, 12,18, 24dB/octave>
  If bypass is selected then a full range unfiltered signal is sent to the speaker.
- **lpf slope** Set the low pass filter slope. Options are <off><6, 12,18, 24dB/octave>
  If <off> is selected then no signal is sent to the sub woofer.
- **sub mode** Push the VOLUME encoder to adjust the sub output mode <mono, left, right, LFE only, mono sub, left sub, right sub>
  - **mono** sum of all lpf outputs and LFE
  - **left** sum of all left and center lpf outputs and LFE
  - **right** sum of all right and center lpf outputs and LFE
  - **LFE Only** direct LFE signal
  - **mono sub** sum of all lpf outputs without LFE
  - **left sub** sum of all left and center lpf outputs without LFE
  - **right sub** sum of all right and center lpf outputs without LFE

Press the SETUP switch to save and exit Setup Mode.

Note: To bypass the Bass Management processing on all channels see the **bass management bypass** in Monitor Control setup. Also, the User ABC switches can be assigned to bypass the bass management system. See A B C User Switches Setup.

6.3.9   SOLO/MUTE Setup

The 8 SOLO/MUTE switches can be "paged" across 3 pages for a total of 24 controls.

Each switch can be mapped to one or more speaker channel.

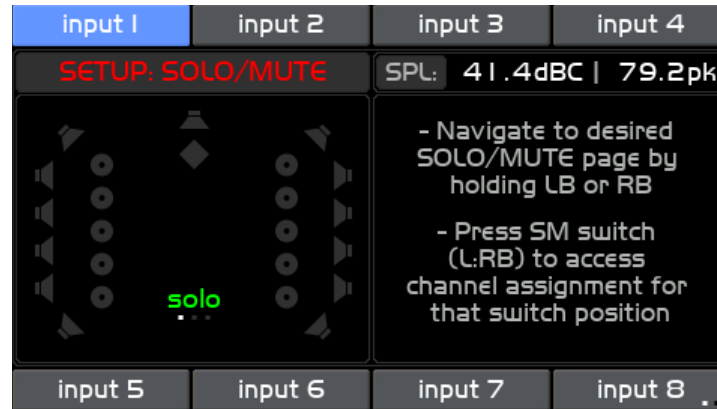Push the SETUP switch then press-hold the SOLO/MUTE switch to enter the Solo/Mute setup

screen.

Illustration 81: Setup SOLO/MUTE Initial Screen

This screen contains a speaker layout area on the left and instructions in the right area.
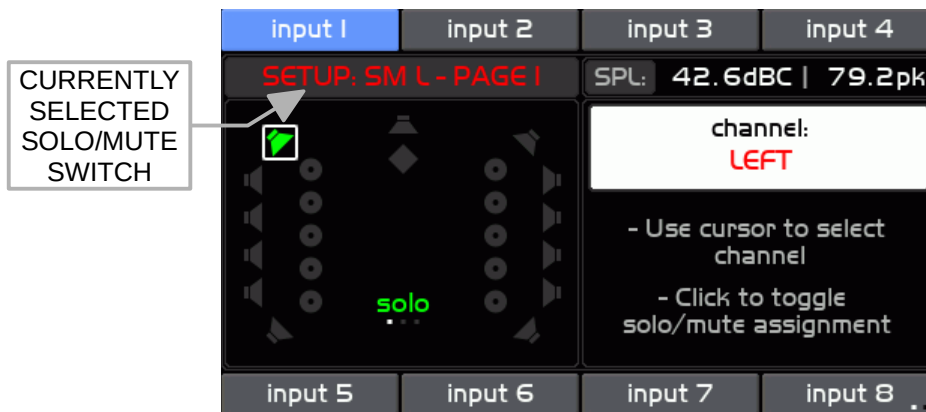

CURRENTLY
SELECTED
SOLO/MUTE
SWITCH

Illustration 82: Setup Solo/Mute

- Push any solo/mute switch (L, C, R, LS, SUB, RS, LB, or RB) and it will enter setup mode for that switch. The text band at the top of left area of the screen will show which solo/mute switch is selected and the speaker (or speakers) that is mapped to the selected switch will be green. Pressing the active switch again activates or de-activates the function and toggles the function on repeated presses.
- The right area of the screen displays name of the currently selected speaker channel. The corresponding channel speaker icon will have a white outline box around it.
- Turn the VOLUME encoder to scroll through the speaker icons in the left side of the screen and click on any speaker icon to assign or de-assign it to the currently selected solo/mute switch.
- Push any other solo/mute switch to repeat the process.

Press-SETUP switch to save and exit.

Note: Since the solo/mute functionality is live during setup it is possible that the currently selected switch will not be illuminated. Be sure to check the switch name in the text band at the top of the left side of the screen.

6.3.10   Room Correction EQ setup

The m908 room correction EQ is accessed through the Control Room Setup screen.

Push the SETUP switch and then any of the CR1, CR2, or CR3 speaker select switches to enter the Setup Speaker screen. Click on **channel setup** to enter the Channel Setup screen. Navigate to **room eq**. The following parameters control the Room Correction EQ

Illustration 83: Speaker EQ Parameter Screen

The Room Correction EQ screen displays 1 to*12* parametric equalizer bands for the selected speaker.

NOTE: As of firmware version 1.0.3 the Room EQ filters are implemented with 32 bit fixed point computations and a 64 bit accumulator. This means that 0dBFS is the maximum internal signal level so using positive EQ values can cause clipping. To avoid this adjust the CR speaker system trim level to a negative value to pre-scale the input to the EQ section. Clipping on a channel will cause the speaker level icon to indicate and *over* condition and flash red. The hold time for the over indication can be adjusted in the System Setup menu.

Each EQ band contains the following parameters:

- **Hz** The center frequency or shelf frequency of the selected filter. Range is off, 10Hz-20kHz with 246 steps in approximately 1/24octave spacing.
- **dB** The boost or cut level for peak, high shelf, and low shelf filter shapes. Range is -12db to +12dB in 0.1dB steps.
- **shape** Selects the filter shape. Options are <low pass, high pass, low shelf, high shelf,

peak>

- **Q** Sets the filter Q for peak filter shape. Range is <0.02 - 50>
- ►Exit band

To navigate the EQ screen:

- Turn the VOLUME encoder to select between the available EQ bands.
- Click the VOLUME encoder to edit an EQ band.
- Turn the VOLUME encoder to select between the available EQ parameters.
- Click the VOLUME encoder to edit the parameters.
- Click the VOLUME encoder again to enter the new value and return to the EQ parameter select mode.
- Turn the VOLUME encoder to select the ► symbol and click to exit and return to EQ band select mode.
- Turn the VOLUME encoder to select Add (+) to add an EQ band to the speaker channel. This will add a band to the end of the list.
- Turn the VOLUME encoder to Subtract (-) to remove an EQ band from the channel. This will remove the last EQ band in the list. Note that when the total number of available EQ bands are used the Add (+) will no longer allow additional bands to be added.
- Turn the VOLUME encoder to select the EXIT bar on the right side of the screen to exit EQ and return to **channel setup.**

6.3.11   Monitor Control setup

Push the SETUP switch then press-hold the VOLUME encoder to enter Monitor Control setup screen.
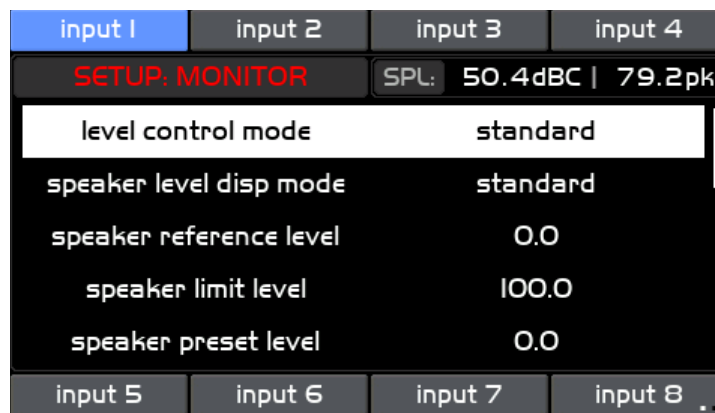

Illustration 84: Setup Monitor Screen

This screen contains the following a list of setup parameters that can be highlighted by turning the VOLUME encoder knob.

NOTE: Headphone related monitor parameters are hidden if Headphone are disabled in the current workflow. The available parameters are:

- **level control mode** Sets the step size for the VOLUME control encoder. Options are <standard, coarse/fine>. In standard mode the VOLUME control adjusts the volume in 0.5dB steps with an acceleration curve that increases the step size the faster the control is rotated. In <course/fine> mode the VOLUME control adjusts the volume in 4.5dB steps for course adjustment. Pushing while turning the VOLUME encoder switches to fine mode and changes the volume in 0.5dB steps.

- **speaker level disp mode** Sets the speaker level display mode. Options are <standard, reference>.

- **speaker reference level** Adjusts the speaker reference level. Range is <0 to 100> in 0.5dB steps. This parameter represents the reference level that is displayed (when speaker level display mode = reference level) at the current volume setting. Available range is the current volume setting +/- 100 in 0.5 steps. This parameter is the same for all speaker outputs.

  The following is an example of how the reference level may be configured for an 82dBSPL reference listening level:

  - Play and monitor a nominal level noise source.
  - Adjust the speaker level until the SPL measurement at the listening position equals 82dBSPL.
  - Enter SETUP then push-hold the VOLUME encoder to enter the Setup Monitor screen and navigate to **speaker reference level** parameter. Adjust it to 82.
  - Set **speaker level disp mode** to <reference>

- **speaker limit level** Sets the maximum volume level that the Control Room speaker can be set to. Range is <0-100> in 0.5dB steps.

- **speaker preset level** Sets the level that the speaker system will go when on push-hold of the VOLUME encoder while in the Home screen. Range is <0-100> in 0.5dB steps.

- **speaker power-up level** Sets the volume level that the speaker system will go to after power up. Range is <0-100> in 0.5dB steps.

- **lfe input lpf freq** Enables and sets the LFE channel low pass filter cutoff frequency. Options are <bypass, 80-120hZ> Step size is 1Hz

- **lfe input lpf slope** Selects the LFE LPF slope. Options are <6, 12,18, 24dB/octave>

- **lfe input gain** Sets the LFE channel gain. Adjustment range is <0db to +10dB> in 0.5dB steps.

- **bass management** Enables or disables bass management. Options are: <bypassed,

enabled>

- **room equalization** Enables or disables all room correction equalizers. Options are: <bypassed, enabled>
- **headphone source** Selects the stereo signal source for the headphone amplifier. Options are <left/right, downmix, cue 1-8>.
  - left/right: the headphone source is the same signal feeding Control Room left and right with mono,l-r and mute behavior as configured in setup menus. If downmix is active in the CR Outputs the headphones will receive the downmixed signal.
  - downmix: Inserts the stereo output phones/cue downmixer for monitoring in stereo when the CR Outputs are monitoring in surround.
  - cue 1-8: Routes one of the Cue outputs to the headphone amplifier. When a cue is being monitored on headphones the Cue output routing is still active.
- **headphone gain** Adjusts the gain of the headphone system to match the sensitivity of your headphones. Adjustment range is -20dB to +20dB in 0.5dB steps.
- **headphone balance** Adjusts the Left-Right balance of the headphones. Depending on the direction the VOLUME encoder is turned the Left or Right channel can be attenuated up to 20dB in 0.5dB steps.
- **headphone limit level** Sets the maximum volume level for the headphone amplifier. Adjustment range is 0 to 100 in 0.5dB steps.
- **headphone preset level** Sets the level that the headphones will go to on push-hold of the VOLUME encoder. Adjustment range is 0 to 100 in 0.5dB steps.
- **headphone power-up level** Sets the volume level that the headphones will go to after power up. Adjustment range is 0 to 100 in 0.5dB steps.
- **exclusive output mode** Sets the exclusive output mode of the m908. Options are: <disabled, standard, mute speakers, mute headphones>
  - off: Speaker system and headphones are on all of the time.
  - standard: Headphones are muted when speaker Volume Control is active. Speaker system is muted when the headphone Volume Control is active.
  - mute speakers: Speaker system is muted when the headphone Volume Control is active. Headphones and speakers are on when speaker Volume Control is active.
  - mute headphones: Headphones are muted when speaker Volume Control is active. Headphones and speakers are on when headphone Volume Control is active.
- **phones/cue downmix mode** Selects the stereo output downmix template for Headphones and Cue outputs. Options are: <7.1>LtRt, 7.1>LtRtPLII, 7.1>LoRo, 5.1>LtRt, 5.1>LtRtPLII, 5.1>LoRo, LCRS>LR, LCRS2>LR>
- **center to left/right** Adjusts the center mix in the Left and Right outputs. Range is: <off, -6.0dB-+3.0dB> Step size is 1.5dB. Parameter is available only when non-PLII

downmix template is selected.

- **surround to left/right** Adjusts the surround mix in the Left and Right outputs. Range is: <off, -6.0dB-+3.0dB> Step size is 1.5dB. Parameter is available only when non-PLII downmix template is selected.

Push the SETUP switch to save and exit Setup Mode.

6.3.12   MONO setup

Push the SETUP switch then the MONO / (L-R) switch to enter the Setup Mono screen. If MONO was active before pushing SETUP it will remain active. If MONO was not active then press the MONO switch to activate the MONO function for auditioning.
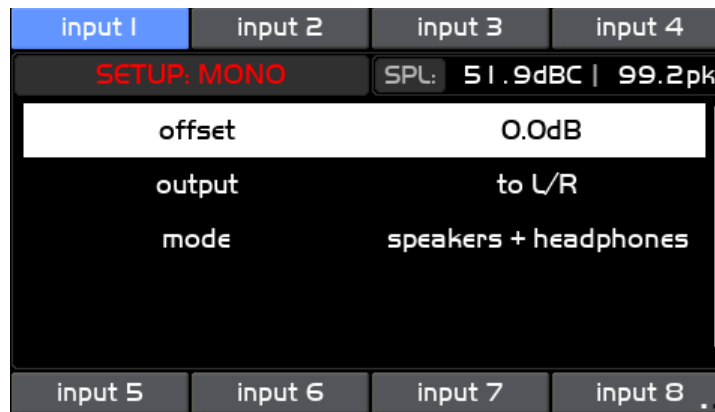
Illustration 85: Setup Mono Screen

This screen contains the following a list of setup parameters that can be highlighted by turning the VOLUME encoder knob. The available parameters are:

- **offset** Sets the amount of attenuation upon summing the Left and Right channels. Range is <0 to -12dB> in 0.5dB steps.

  Analog mixers would have varying summing laws for summing to mono depending on the type of circuitry being used. Passive summing would result in a -6dB attenuation for signals panned hard right or left. Active summing amplifiers would result in a -3dB attenuation. Use this parameter to match the perceived loudness from stereo to mono.

- **output** Sets the output path for the MONO signal. Options are<to L/R, to C>
  - ○ to L/R sends the MONO signal to the Left and Right speaker equally.
  - ○ to C Sends the MONO signal to the Center speaker only. Left and Right speakers are muted.

- **mode** Sets which monitor paths are affected by MONO. Options are <speakers +

headphones, speakers only, headphones only>
Press the SETUP switch to save and exit Setup Mode.

6.3.13   (L-R) setup

Push the SETUP switch then push-hold MONO / (L-R) switch to enter the Setup (L-R) screen. If (L-R) was active before pushing SETUP it will remain active. If (L-R) was not active then press-hold the (L-R) switch to activate the (L-R) function for auditioning
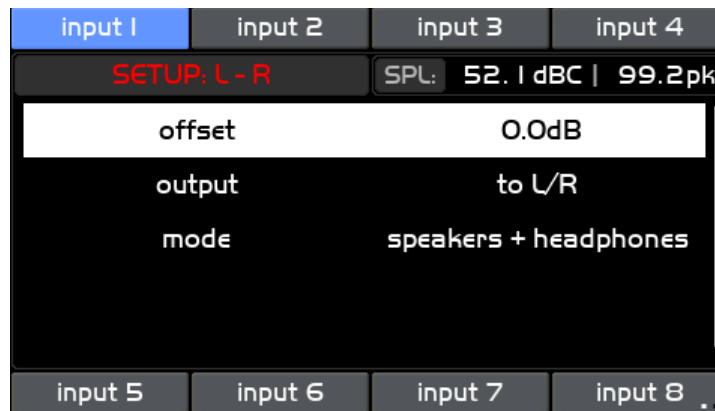

Illustration 86: Setup L-R Screen

This screen contains the following list of setup parameters that can be highlighted by turning the VOLUME encoder knob. The available parameters are:

- **offset** Sets the amount of gain applied upon subtracting the Right from Left channel. This parameter can be adjusted from 0 to +6dB in 0.5dB steps. Use it to match the perceived loudness from stereo to L-R.
- **output** Sets the output path for the L-R signal. Options are<to L/R, to C>
  - ○ to L/R sends the L-R signal to the Left and Right speaker equally.
  - ○ to C Sends the L-R signal to the Center speaker only. Left and Right speakers are muted.
- **mode** Sets which monitor paths are affected by the L-R processor. Options are <speakers + headphones, speakers only, headphones only>

Push the SETUP switch to save and exit Setup Mode.

6.3.14   CUE setup

Push the SETUP then push the MON>CUE switch to enter the Setup Cue Outs screen. NOTE: the current state of mon>cue will be unchanged by this action. Once in Cue Outs setup, the MON>CUE switch returns to normal operation.
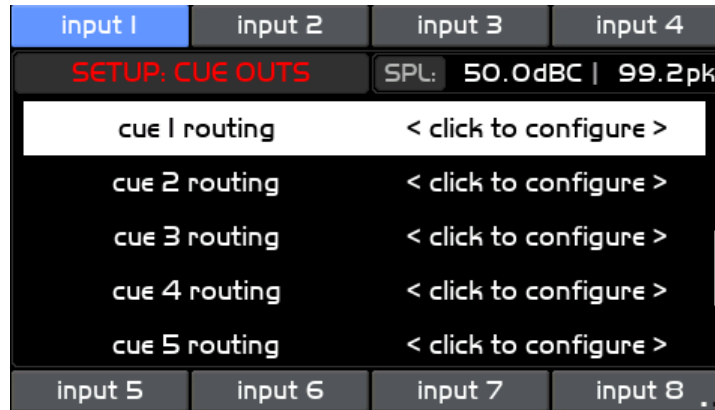
Illustration 87: Setup Cue Outs Screen

This screen contains the following list of setup parameters that can be highlighted by turning the VOLUME encoder knob. The list of parameters will vary depending on how many Cue channels are enabled in the current Workflow. The available parameter list below will repeat for each Cue path in the current Workflow:

- **cue X level** 0-100 This parameter adjusts the output level of the selected CUE send. Volume range is <0 to 100> in 0.5dB increments. 0 is muted and 100 is unity gain.
- **cue X routing** Clicking on this will enter the cue routing configuration screen. This is where CUE paths can be assigned to hardware inputs and outputs.
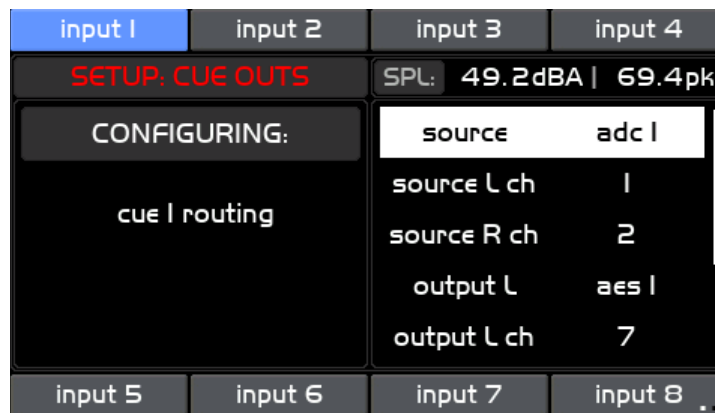
Illustration 88: Setup Cue Routing Screen

The Cue Outs routing screen will display the current cue number selected for editing in the left side window. The right window contains the list of cue configuration parameters:

- ○ **source L** Selects the left channel source connector. Scroll through this list and choose the desired input connector.

- ○ **source L ch** Selects the left channel source connector channel. Scroll through this list and choose the desired input connector channel.
- ○ **source R** Selects the right channel source connector. Scroll through this list and choose the desired input connector.
- ○ **source R ch** Selects the right channel source connector channel. Scroll through this list and choose the desired input connector channel.
- ○ **output L** Selects the left channel output connector. Scroll through this list and choose the desired output connector.
- ○ **output L ch** Selects the left channel output connector channel. Scroll through this list and choose the desired output connector channel.
- ○ **output R** Selects the right channel output connector.
- ○ **output R ch** Selects the right channel output connector channel.
- • **mon>cue**   controls mon>cue function for this cue output.
  - ○ disabled: the cue source is always the cue input regardless of the state of MON>CUE
  - ○ left / right: When MON>CUE is activated, the cue source is switched to the left and right channels of the selected Control Room monitor source.
  - ○ downmix: When MON>CUE is activated, the cue source is switched to a stereo downmix of the selected Control Room monitor source.

NOTE: To use the headphones for monitoring a Cue path when headphones are not enabled in the system workflow simply route the Cue outputs to CUE/CR EXT channels 7 and 8. The DAC channels that feed this connector also feed the headphone amplifier. The headphone volume is controlled by the Cue level in Setup>Cue.

The User ABC switches can be assigned to mute the *input* of a CUE path. This allows the talkback signal and MON>CUE signals to still be output but the source signal will be muted. See A B C User Switches Setup.

Push the SETUP switch to save and exit Setup Mode.

6.3.15   MUTE setup

Push the SETUP switch then the MUTE switch to enter the Setup Mute screen.If MUTE was active before pushing SETUP it will remain active. If MUTE was not active then press the MUTE switch to activate the MUTE function for auditioning.
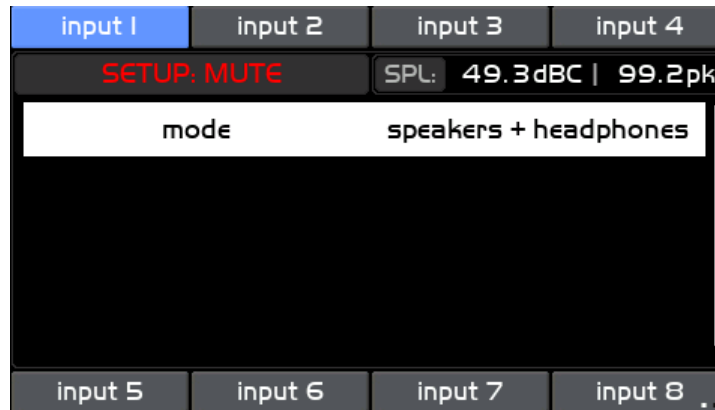
Illustration 89: Setup Mute Screen

This screen contains the following setup parameter that can be adjusted by pushing the VOLUME encoder knob.

- **mode** The Mute control can be enabled for the following monitor paths:

    speakers + headphones

    speakers only

    headphones only

Push the SETUP switch to save and exit Setup Mode.

6.3.16 User A B C Switches Setup

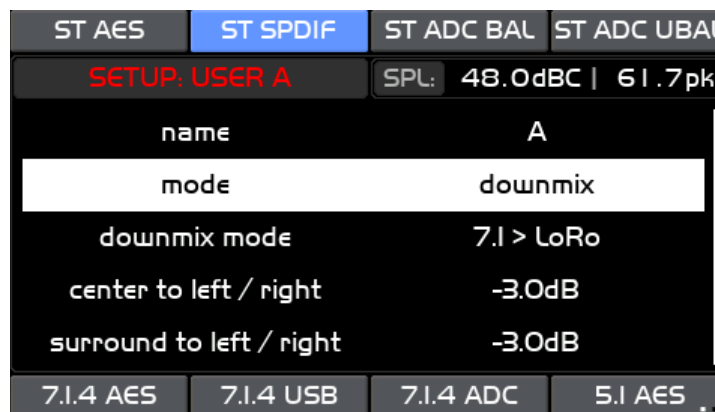Push the SETUP switch then push the A, B, or C User switch to enter the Setup User switch screen.

Illustration 90: Setup ABC User Switch Screen

This screen contains the following setup parameters:

- **name** Use the Standard Name Editing screen to name the special User switch. Create a name that describes the use of the switch. i.e. Downmix 5.1
- **mode** Sets the User switch mode. Options are <disabled, downmix, talkback, gpio, bass man bypass, cue input 1-8 mute>
  - <downmix> Sets the User switch mode to downmix.
  - <talkback> Sets the User switch mode to talkback. From the Talkback setup menus the switch can be set to control which talkback mic is activated and which Cue sends the mic is routed to. When configured for <talkback> the User switch will illuminate RED when active.
  - <gpio> Sets the User switch mode to gpio. When set for gpio the switch can trigger an output event on one of the gpio pins. Go to System Setup to map a gpio pin to the User switches.
  - <bass management bypass> Sets the User switch mode to bypass the bass management system. This will send full range signals to all speakers and mute the sub woofer low pass signals.
  - <CUE in mute 1..8> Sets the User switch mode to mute the input of any one of the defined CUE paths. This will not mute the CUE output so Talkback signals and MON>CUE is still functional.
- **downmix mode** Sets the Downmix template. Shown when **mode** is <downmix> See the Downmix Operation section of this manual for details on downmix templates.
- **center to left/right** Sets the mix level of the Center channel to the Left and Right channel speakers. Available for stereo output downmixes. Options are <off, -6dB to +3db in 1.5dB steps>
- **surround to left/right** Set the mix level of the Surround channels to the Left and Right channel speakers. Available for stereo output downmixes. Options are <off, -6dB to +3db in 1.5dB steps> Available in all non-PLII Stereo downmixes.
- **lfe** Enables or disables the lfe channel. Options are <disabled, enabled>. Available on 7.1>5.1 and 7.1>5.1PLII

6.3.17   Downmix setup

Downmixing is included in the A B C User Switches Setup.

6.3.18   Meter Output setup

Push the SETUP switch then push-hold the MUTE switch to enter the Setup System screen.

Turn the VOLUME encoder to scroll to and click on **meter output source**

The list of available sources will include all of the system inputs <INPUT 1-16, input monitor

pre, input monitor post>.

- ○ <INPUT 1-16> refers to defined INPUT sources for the m908. Choosing one of the inputs will fix the Meter outputs to the selected input. There will be no changes to the signal level.
- ○ <input monitor pre> Selecting this will send whatever source is being monitored to the Meter output. The signal level will follow the Input Level Offset and Input Summing.
- ○ <input monitor post> Selecting this will send whatever source is being monitored to the Meter output. The meter signal will have all speaker processing applied: volume, delay, offset, bass management, and room EQ.

Once an input source for the meter outputs has been selected turn the VOLUME encoder to scroll to **meter output routing** and click to enter the Meter Output setup screen.
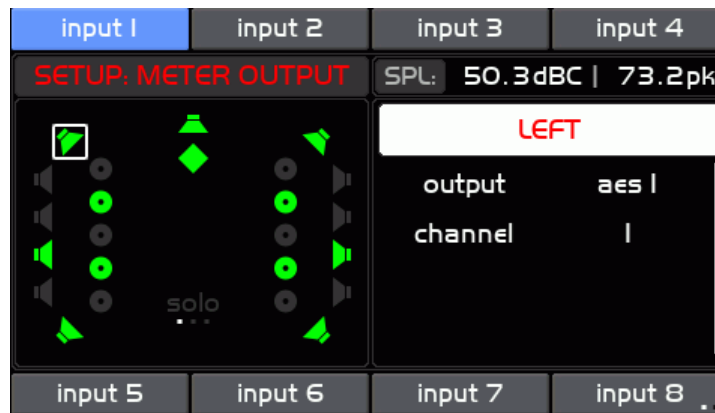
Illustration 91: Meter Output Setup Screen

The Meter Output Setup screen is where the meter source signals are routed to physical output connectors.

The left side of the screen window shows all of the available signals enabled in the current Workflow. Speaker Icons that are green are enabled for the meter output. Speakers that are disabled are gray. The currently selected channel has a white box around it.

On the right side of this screen there is a list of options and the top item is highlighted initially:

- ○ **channel name** The name of the currently currently selected channel in the left window. Pushing the VOLUME encoder enters the Channel Select mode. Turning the encoder scrolls through the available channels. Pushing the VOLUME encoder selects the highlighted channel for routing. See the Standard Channel Routing section of this manual for details on channel routing.

- ◦ **output** Selects the meter output connector for the currently selected speaker. Select "disabled" if the selected channel is not used in the meter output. The selected speaker icon will turn gray.
- ◦ **channel** Select the desired connector channel for the currently selected DSP channel. Select "disabled" if the selected channel is not used in the meter output. The selected speaker icon will turn gray.

NOTE: If the selected meter output connector channel is already in use for CUE or Control Room speaker signals a dialog box will appear asking if you want to replace the existing routing with the new selection.

NOTE: Meter outputs can not go to the analog CR1, CR2, and CR3 OUT connectors. Meter outputs can be routed to CUE/CR EXT. output connector.

To exit the Meter Output Setup screen and return to the System Setup screen press the MUTE switch.

6.3.19   Talkback setup

Push the SETUP switch then the TALKBACK switch to enter the Setup Talkback screen.

NOTE: the current state of talkback will be unchanged by this action. Once in Talkback setup, the TALKBACK switch returns to normal operation.

Illustration 92: Setup Talkback Screen

This screen contains the following setup parameters that can be adjusted by turning the VOLUME encoder knob.

- • **rcu mic gain** Sets the RCU built in microphone gain. Range is <0-30dB> in 1dB steps.
- • **ext mic gain** Sets the gain of the ACU microphone preamplifier. Range is <+11, +19-74dB> in 1dB steps.
- • **ext mic +48V** Enables or disables +48V phantom power for the ACU talkback microphone amplifier.

- **talkback output** Sets the signal source for the Talkback Output XLR connector. Options are <follow talkback, rcu mic, ext mic, ext mic fixed> When set for <follow talkback> the XLR will output the signal from which ever talkback mic is active. In the case where more than one talkback mic are active the first mic to be active will be output. **Note:** This output is only active when Talkback is active. When <ext mic fixed> is selected the external mic signal will always be present on the XLR talkback output.
- **monitor dim mode** Sets which monitor paths are dimmed when talkback is active. Options are <disabled, speakers + headphones, speakers only, headphones only>
- **monitor dim level** Sets the dim level when talkback is active. Range is <0 to -30dB> in 0.5dB steps.
- **cue dim** Enables or disables Cue dimming when talkback is active.
- **cue dim level** Sets the dim level on all CUE paths that have dimming enabled when talkback is active. Range is <0 to -30dB> in 0.5dB steps.
- **activation mode** Sets the RCU and external talkback switch activation mode. Options are <auto, momentary, latching> This parameter also affects how the User ABC switches operate if they are assigned to Talkback mode.
- **rcu switch mic assign** Sets which talkback mic is activated by the RCU TALKBACK switch. Options are <rcu mic, ext mic>
- **rcu switch cue assign** Selects Cue paths to route the talkback signal to when the RCU TALKBACK switch is pushed. Clicking on this option opens the Setup Talkback RCU Switch Cue Assign screen.
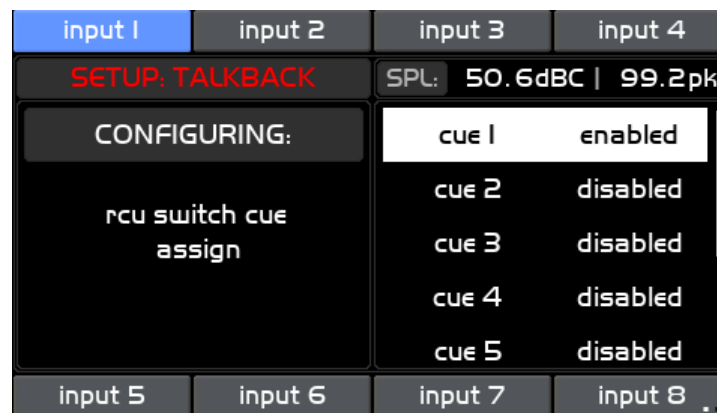
Illustration 93: Setup Talkback Switch Assign Screen

The left window of this screen indicates which talkback switch is being configured. The right side of the screen lists the available Cue paths based on the System Workflow.

- **cue x** Enables or disables talkback send to the currently selected cue path.
- **EXIT** Returns to the previous Setup Talkback screen.

- **ext switch mic assign** Sets which talkback mic is activated by the ACU External Talkback Mic Switch. Options are <rcu mic, ext mic>
- **ext switch cue assign** Selects Cue paths to route the talkback signal to when the External Talkback switch is pushed. Clicking on this option opens the Setup Talkback External Switch Cue Assign screen. This screen is identical to the Setup Talkback RCU Switch Cue Assign screen described above.
- **ABC switch mic assign** Sets which talkback mic is activated by the A, B, or C User switch. If the A, B, or C User switches have been configured to be in Talkback mode they will show up in the list. Options are <rcu mic only, ext mic only, rcu and ext. mic>
- **ABC switch cue assign** Selects Cue paths to route the talkback signal to when the A, B, or C User switch is pushed. If the A, B, or C User switches have been configured to be in Talkback mode they will show up in the list. Clicking on this option opens the Setup A, B, of C Switch Cue Assign screen. This screen is identical to the Setup Talkback RCU Switch Cue Assign screen described above.
- **gpio 1234 switch mic assign** Sets which talkback mic is activated by the GPIO 1, 2, 3, or 4 external switch. If any of the GPIO 1, 2, 3, or4 pins have been configured for Talkback mode they will be listed here. Options are <rcu mic only, ext mic only, rcu and ext. mic>
- **gpio 1234 cue assign** Selects Cue paths to route the talkback signal to when the GPIO 1, 2, 3, or 4 external switches are closed. If any of the GPIO 1, 2, 3. or 4 pins have been configured for Talkback mode they will be listed here. Clicking on this option opens the Setup GPIO X Cue Assign screen. This screen is identical to the Setup Talkback RCU Switch Cue Assign screen described above.

6.3.20   SPL Meter setup

Push the SETUP switch then push-hold the MON>CUE (spl clear) switch to enter the Setup SPL Meter screen.

Illustration 94: Setup SPL Meter Screen

This screen contains the following setup parameter that can be adjusted by pushing the VOLUME encoder knob.

- **meter** This parameter enables or disables the SPL meter.
- **filter** Selects the weighting filter for the SPL measurement. Options are:
  none: No filter applied.
  a-weighted: A weighting filter applied.
  c-weighted: C weighted filter applied
- **peak mode** Sets the behavior of the peak SPL measurement. Options are:
  off: Peak values are not saved. The peak SPL display value will follow the real time SPL display value.
  fast: The peak SPL value is held for about 3 seconds.
  med: The peak SPL value is held for about 6 seconds
  slow: The peak SPL value is held for about 12 seconds
  hold: The peak SPL value is held until manually reset with a push-hold of the MON>CUE (spl clear) switch.
- **response** Sets the response averaging time of the SPL meter. Options are:
- slow: SPL meter uses a slow average.
  fast: SPL meter uses a fast average.

6.3.21 Clock Source Setup

Push the SETUP switch then push-hold the MUTE switch to enter the Setup System screen.
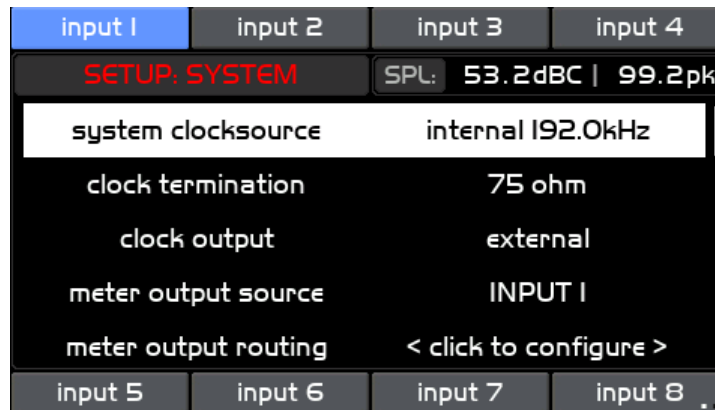
Illustration 95: Setup System Screen

Use the VOLUME encoder knob scroll to **clocksource** and click to select the system clock source. Options are <internal 44.1kHz, internal 48kHz, internal 88.2kHz, internal 96.0kHz, internal 176.4kHz, internal 192.0kHz, wordclock, option io, aes-1, aes-2, aes-3, aes-4, optical-1, optical-2, usb, spdif, dante, digilink, ravenna>

- ○ internal Sets the clock source to be the internal clock generator in the m908.
- ○ wordclock Sets the clock source to the Word Clock input. The system will synchronize to the incoming word clock. All inputs to be monitored must be at the same sample rate and synchronous to the word clock.
- ○ usb Sets the clock source to USB. The m908 USB audio interface operates in *asynchronous* mode which means that the m908 is the clock master and is operating on the internal clock. However, the USB host determines the system sample rate.
- ○ option io Sets the clock source to the Option IO module slot. This can be Dante, DigiLink, , or AES67/Revenna. If using a DigiLink module the clock source will be internal and the DigiLink module determines the system sample rate. The system will slave to all other types of IO modules.
- ○ aes-1-3 Sets one of the AES3 inputs as the clock source. AES3 1-3 are 8 channel interfaces which each contain 4 AES3 signals. The system will synchronize to the lowest numbered valid signal. All four signals must be at the same sample rate and must be synchronous to each other.
- ○ aes-4 Sets the aes-4 (XLR) as the clock source.
- ○ spdif Sets the clock source to the SPDIF coaxial input.
- ○ optical-1 Sets the clock source to the optical connector 1 (can be ADAT or TOSLINK)
- ○ optical-2 Sets the clock source to the optical connector 2 (can be ADAT or TOSLINK)

6.3.22   Word Clock Termination Setup

Push the SETUP switch then push-hold the MUTE switch to enter the Setup System screen.

Turn the VOLUME encoder to scroll to and click on **clock termination**

Options are <48kΩ , 75Ω>.

Push the Setup switch to save changes and exit Setup Mode.

6.3.23   Word Clock Output Setup

Push the SETUP switch then push-hold the MUTE switch to enter the Setup System screen.

Turn the VOLUME encoder to scroll to and click on **clock output**

Options are <external, system clock,>

- ◦ <external> the signal on the word clock output BNC connector will be a buffered copy of the signal on the word clock input BNC.
- ◦ <system clock> the signal on the word clock output BNC connector will be the internal system clock source.

  Note: If the system clock source is set to word clock and the word clock output is set to system clock, then the output clock signal will be a re-clocked version of the input clock.

Push the Setup switch to save changes and exit Setup Mode.

6.3.24   GPIO Setup

The m908 has 4 GPIO (General Purpose Input/Output) pins on various rear panel connectors. These pins can be configured as logic level inputs or outputs and can be mapped to various functions. The GPIO pins can be programmed to operate at 3.3V or 5.0V by moving jumpers on the PCB. Built in pull up resistors allow for the use of external switch closures to trigger input events.

GPIO configuration is accessed in the System Setup screen.

The table below contains the available input and output functions as well as the logic mode.

GPIO Functions			
MODE	FUNCTION	Description	Trigger
INPUT	TALKBACK	Activates TALKBACK	logic level
INPUT	User Switch A	Activates User Switch A	logic level
INPUT	User Switch B	Activates User Switch B	logic level
INPUT	User Switch C	Activates User Switch C	logic level
INPUT	Input 1-16	Selects Monitoring Input 1-16	edge
INPUT	CR1-3	Selects Control Room Speaker System CR1, CR2, or CR3	edge
INPUT	MUTE	Activates MUTE function	logic level
INPUT	DIM	Activates DIM function	logic level
INPUT	MONO	Activates MONO function	logic level
INPUT	L-R	Activates L-R (left minus right) function	logic level
INPUT	MON>CUE	Activates MON>CUE function	logic level
OUTPUT	TALKBACK RCU	TALKBACK RCU tally	
OUTPUT	TALKBACK EXT	TALKBACK EXT tally	
OUTPUT	User Switch A	User Switch A tally	
OUTPUT	User Switch B	User Switch B tally	
OUTPUT	User Switch C	User Switch C tally	
OUTPUT	Input 1-16	Monitoring Input 1-16 tally	
OUTPUT	CR1, CR2, CR3	Control Room Speaker System CR1, CR2, or CR3 tally	
OUTPUT	MUTE	MUTE function tally	
OUTPUT	DIM	DIM function tally	
OUTPUT	MONO	MONO function tally	
OUTPUT	L-R	L-R (left minus right) function tally	
OUTPUT	MON>CUE	MON>CUE function tally	

When assigning a GPI to a monitoring function the state of the GPI will be "OR"ed with the state of the corresponding RCU switch. For instance, if a GPI is assigned to MUTE, the mute function will activate from the GPI **or** the RCU. If the function is being activated from GPI the RCU switch will flash. If the MUTE was already activated from the RCU then MUTE will remain active until the GPI is disabled and the MUTE switch is pressed.

The table and illustration below indicates the locations of each GPIO pin and the circuit board jumper for selecting the logic level for each pin. See the m908 Electrical Specifications to make

sure the GPIO electrical characteristics are compatible with your external equipment.

Before adjusting the GPIO pcb jumpers turn of the power at the PSU and remove the AC power cord from the m908 PSU. Remove the m908 ACU top cover using a #2 Phillips screwdriver.

There are ESD sensitive components on the circuit boards inside the m908. Take care to insure that your work space is free of static electricity before removing the cover of the m908 ACU.

GPIO Voltage Configuration (Factory Default is 5V)		
GPIO#	CONNECTOR	Voltage Select Jumper +3.3V/+5.0V
1	AES3 1	J45
2	AES3 2	J46
3	AES3 3	J47
4	RCU DB15	J50
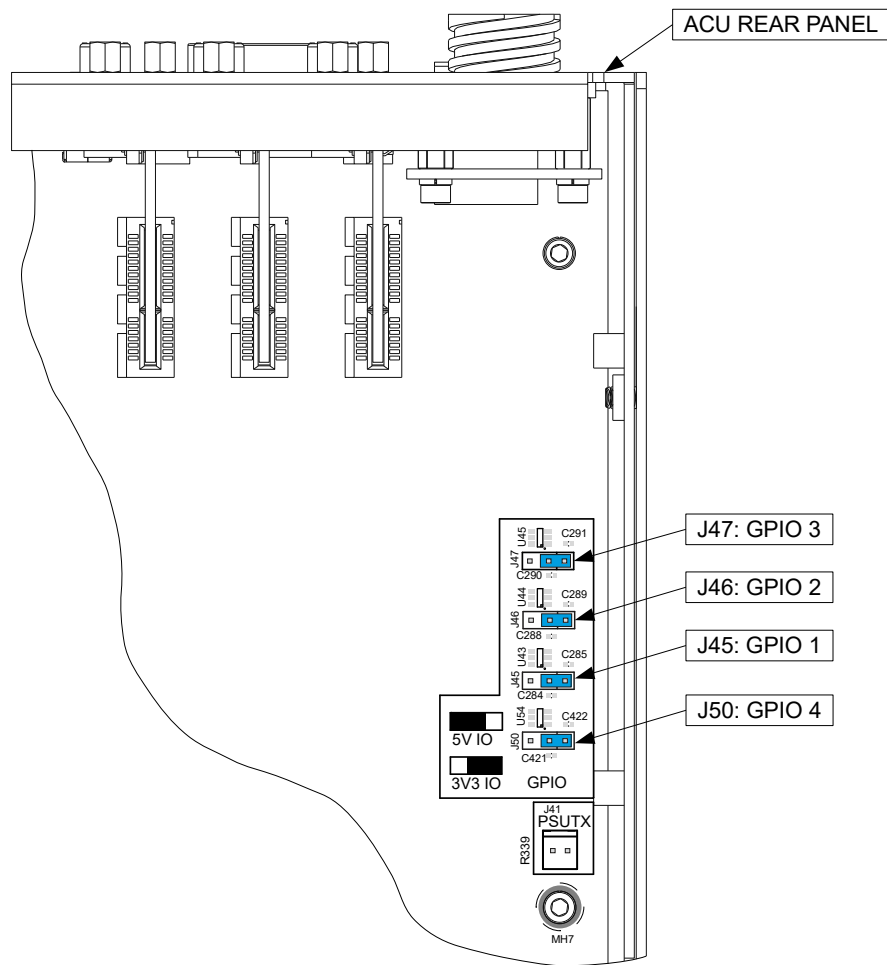
Illustration 96: GPIO Jumper Locations
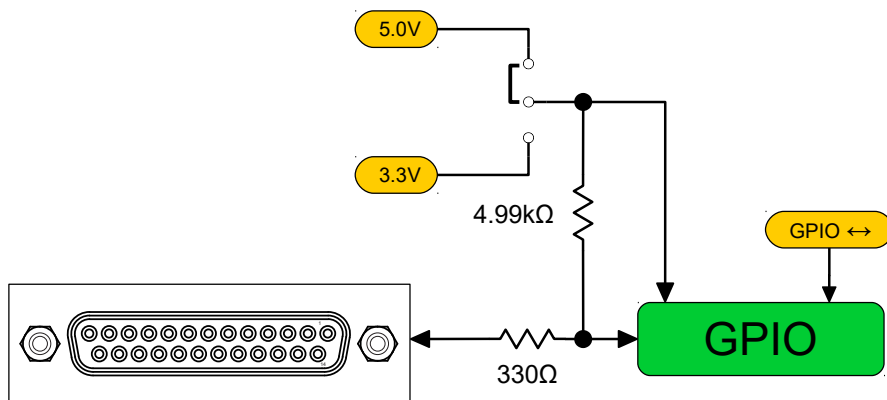
Illustration 97: GPIO Schematic Diagram

6.3.25   System Setup

Push the SETUP switch then push-hold the MUTE switch to enter the Setup System screen.
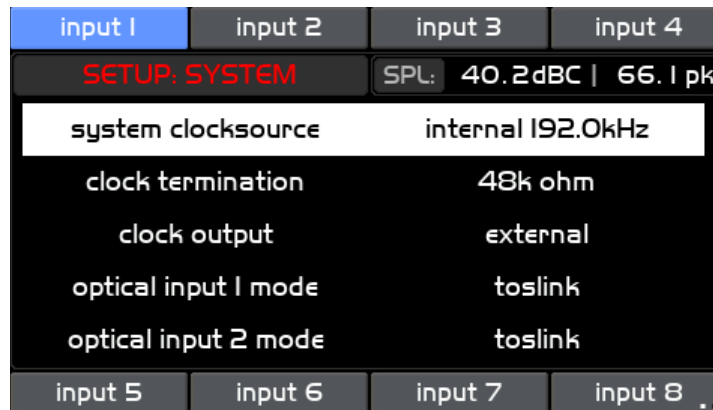
Illustration 98: Setup System Screen

This screen contains the following setup parameter that can be adjusted by pushing the VOLUME encoder knob.

- **clock source** Sets the clock source for the m908 system clock. Options include <internal, word, usb, option io, aes1-4, spdif, adat> See Clock Source Setup above for details.
- **clock termination** Sets the Word Clock input connector termination. Options are <48k ohms, 75 ohms> If the m908 is being fed directly from a word clock source and it is the only receiver then set the termination to 75Ω. If the m908 Word Clock input is being fed in a "daisy-chain" with another terminated receiver then set the termination to 48kΩ.
- **clock output** Sets the Word Clock output signal source. Options are <clock in, system clock> When clock in is selected the Word Clock output buffers and re-transmits the incoming word clock. When set to system clock the Word Clock output transmits the system clock. Use the system clock setting when the m908 is the clock master in a studio.
- **unbal input mode**   Select either unbalanced line level input or RIAA phono level. RIAA Phono preamp option is required. Options are <line, phono>
- **optical input 1 mode**   Sets the mode for optical input connector 1. Options are <TOSLINK, ADAT, SMUX>
  - TOSLINK Stereo PCM. 96kHz maximim sample rate

- ◦ ADAT 8 channel PCM. 44.1kHz or 48kHz sample rate
- ◦ SMUX 4 channel ADAT 88.2kHz or 96kHz sample rate
- **optical input 2 mode**   Sets the mode for optical input connector 2. Options are <TOSLINK, ADAT, SMUX>
  - ◦ TOSLINK Stereo PCM. 96kHz maximim sample rate
  - ◦ ADAT 8 channel PCM. 44.1kHz or 48kHz sample rate
  - ◦ SMUX 4 channel ADAT 88.2kHz or 96kHz sample rate
    Note: when changing the mode of an optical connector the input mode will not be changed until the user clicks the encoder. If the optical mode for the selected connector has changed, any routes associated with it will be disconnected.
- **meter output source** Selects the Meter Output source. See Meter Output setup
- **meter output routing** Enters the Meter Output routing setup screen. See Meter Output setup.
- **backlight** Sets the LCD display back-light intensity level. Options are <min, med, high, max> Adjust the back-light to the lowest level that is acceptable bright for your work environment.
- **display dim** Sets the amount of idle time before the display automatically dims. Options are <off, 10, 15, 20, 25, 30, 45, 60 minutes>.
  NOTE: For long display back-light life it is highly recommended to set the display dim to activate. The backlight dim timer is restarted after any user input on the RCU. If the timer reaches it's preset time the display will dim. Pushing any of the switches or turning the encoder knob will bring the backlight back to its normal brightness.
- **gpio x mode** Sets the mode of the GPIO 1, 2, 3, and 4 pins. Options are <disabled, input, output> There is a gpio mode parameter for each of the 4 GPIO pins.
- **gpio x logic** Set the logic level for the GPIO 1, 2. 3, or 4 pins. Options are <active low, active high> There is a gpio logic parameter for each of the 4 GPIO pins. For logic driven input functions, the active logic state turns the function on, while the opposite state turns the function off. For edge triggered input functions the active low setting selects a falling edge trigger and the active high setting selects a rising edge trigger.
- **gpio x input function** Sets the function for the selected GPIO 1, 2, 3, or 4 pins when in <input> mode. Options are <disabled, talkback, A, B, C, input 1-16, speaker 1-3, mute, dim, mono, l-r, mon>cue> For any of these functions, a change on the RCU will override the GPI setting. Input functions 1-16 and speaker 1-3 functions are edge triggered. All other input functions are logic driven.
- **gpio x output function** Sets the function for the selected GPIO 1, 2, 3, or 4 pins when in <output> mode. Options are <disabled, talkback RCU switch, talkback EXT switch, A, B, C, input 1-16, speaker 1-3, mute, dim, mono, l-r, mon>cue>

- **tcp/ip mode** Selects Ethernet port address mode. Options are <dhcp, static> In dhcp mode the m908 Ethernet port will be assigned an ip address automatically by a DHCP server. If your network does not have a DHCP server then select <static>. The ip address and subnet mask will need to be entered manually using the parameters below.

- **subnet mask** In DHCP mode displays the current subnet mask for the ACU Ethernet port. In static ip address mode this parameter can be edited for the desired subnet mask value.
- **ip address** In DHCP mode displays the current IP address on the ACU Ethernet port. In static ip address mode this parameter can be edited for the desired ip address.
- **mac address** Displays the MAC address of the ACU Ethernet port.
- **rcu firmware version** Displays the current RCU firmware version.
- **acu firmware version** Displays the current ACU firmware version.
- **aux firmware version** Displays the current AUX module firmware version, if available.
- **meter over-hold mode** Sets the speaker icon over (red) indication hold time. Options are: <off, 2 sec, 10 sec>
- **password** Sets the system password which is used to gain access to the web UI. Use the Standard Name Editing screen to name the special User switch. Passwords can be up to 12 characters. The default password is set to: grace

7   Option Modules

7.1 ADC Option Module

The m908 can accommodate up to 2 8-channel ADC modules. The ADC accepts 8 channels of balanced audio on a DB25 connector and also two channels of unbalanced analog on RCA phone jacks and two channels of balanced analog on XLR connectors. The RCA and XLR connectors are relay selected as alternate sources to channels 1-2 of the ADC. If one ADC option is installed in the m908 then the two stereo connectors are installed as well. The second ADC option module only has 8 channels in on DB25.

The stereo inputs on XLR and RCA jacks route to ADC channels 1 and 2. If an input is defined to use these connectors the ADC1 channels 1-2 can not be used at the same time. The illustration below shows how the stereo and 8 channel ADC input is implemented in the m908.
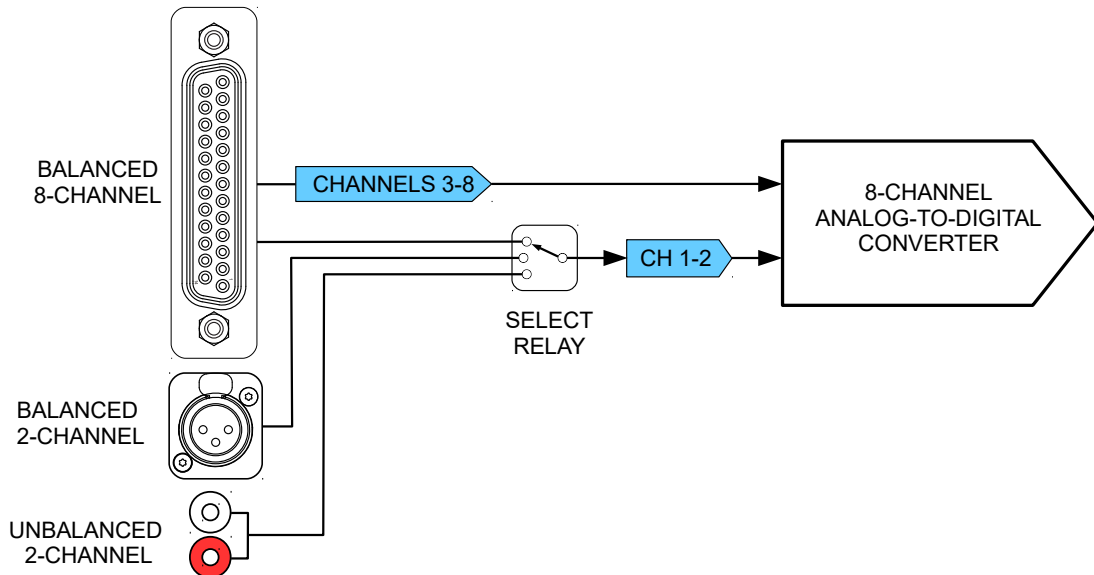
Illustration 99: ADC Stereo Input Routing

If an input is using one of these connectors and you attempt to route another channel to the same connector the following conflict dialog box will appear:

Illustration 100: ADC Input Conflict

Illustration 101: ADC Jumper Locations

7.2 Dante Option Module

The Dante option provides the m908 with two gigabit Ethernet ports for streaming up to 32 channels of audio from and to other Dante enabled devices. This option is available on new and existing units. Please contact your Grace Design dealer or distributor for details. The heart of the Dante interface is the Brooklyn II module. This is where the digital audio data is sent to and received from the Dante network "flows".
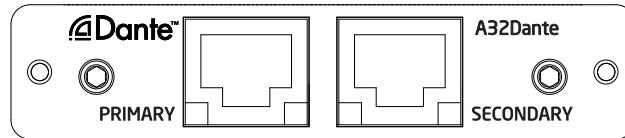
Illustration 102: Dante Option Module

Dante Channel Count		
Sample Rate	Input Channels	Ouptut Channels
44.1-48kHz	32	32
88.2-96kHz	32	32
176.4-192kHz	16	16

We strongly recommend reading through the Dante Controller software user guide to become familiar with Dante networking basics.

The pdf can be downloaded at https://www.audinate.com/resources/technical-documentation

7.3 Ravenna/AES67 Option Module

The Ravenna/AES67 option provides the m908 with two gigabit Ethernet ports for streaming up to 32 channels of audio from and to other Ravenna or AES67 enabled devices. This option is available on new and existing units. Please contact your Grace Design dealer or distributor for details. The Ravenna interface contains the Merging Audio ZMAN module. This allows the m908 to integrate seamlessly in to existing Merging Ravenna systems. Also, being AES67/SMPTE ST2110 compliant, this option will allow for connections to other Ethernet audio interfaces such as DANTE in AES67 mode.

The Ravenna/AES67 Option Module can stream 32 channels in and out at sample rates up to 192kHz.
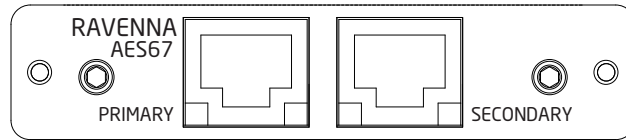
Illustration 103: Ravenna/AES67 Option Module

7.4 DigiLink Option Module

The m908 Digilink option module provides 32 channels of I/O from a Pro|Tools HDX or Pro|
Tools HD Native system. The m908 will emulate two Avid HD I/O interfaces. This option is
available on new and existing units. Please contact your Grace Design dealer or distributor for
details.
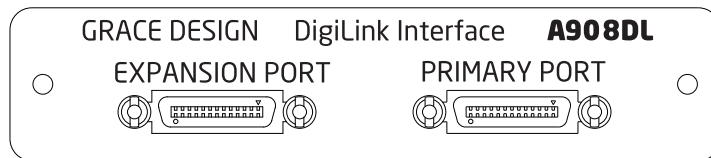
Illustration 104: Digilink Option Module

The m908 emulates *two* HD I/O interfaces with a total of 32 channels of I/O. The
EXPANSION port is currently not active.

In Pro|Tools Hardware Setup the only control that is active for the emulated HD I/Os is the
Clock Source selection. Valid options are Internal and Word Clock. If choosing Word Clock
select the option that is at the session sample rate. The m908 does not support Loop Sync.

If the m908 is the only device connected to the HDX port then the Clock Source selection is
only valid for the HD I/O #1. HD I/O #2 is internally synchronized inside the m908 Digilink
card.

When using the m908 Digilink interface System Clock source or Input Clock Override should
be set to "Digilink". When Word Clock is selected clock source in Pro|Tools Hardware Setup the
m908 Digilink clock will switch to the Word Clock Input BNC connector on the ACU
automatically. You do not need to change the m908 clock source to Wordclock.

Although the m908 will appear as two HD I/O devices with 32 *inputs* these channels are not
currently routable in the m908. The m908 Digilink interface is for monitoring outputs only at this

time. Future m908 firmware will allow analog or digital inputs on the m908 to be routed to the Digilink interface.

There are two ways to connect the m908 to the HDX card or HD Native interface. See the illustrations below.
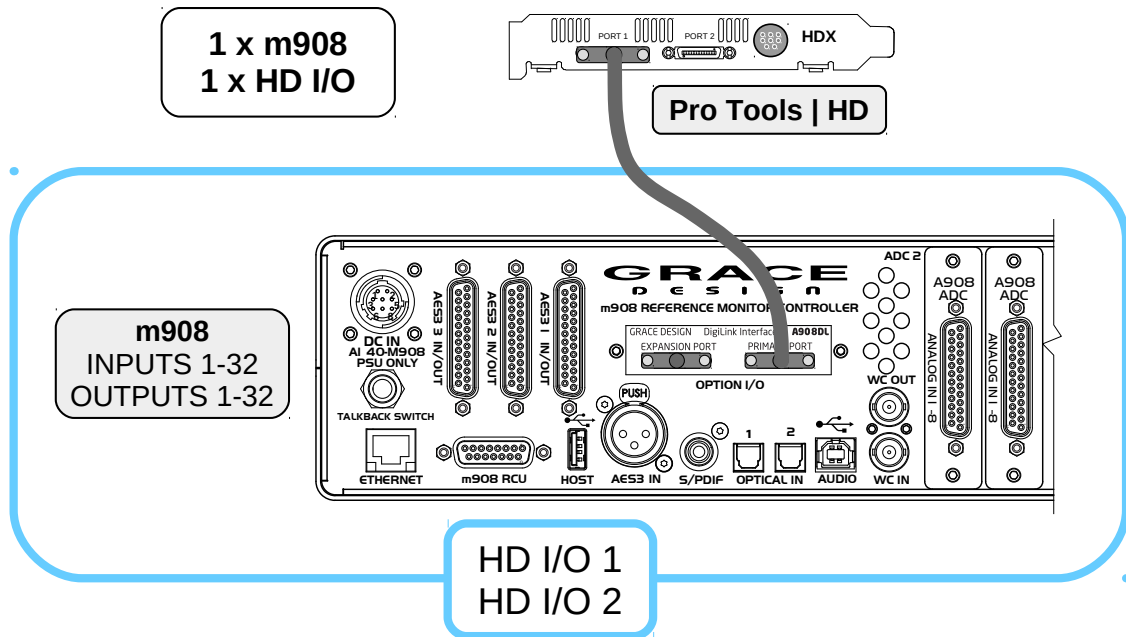
Illustration 105: 32 Channel Digilink Connections

When connected directly to an HDX or Native port the m908 will appear as two HD I/O interfaces and will allow monitoring 32 channels.
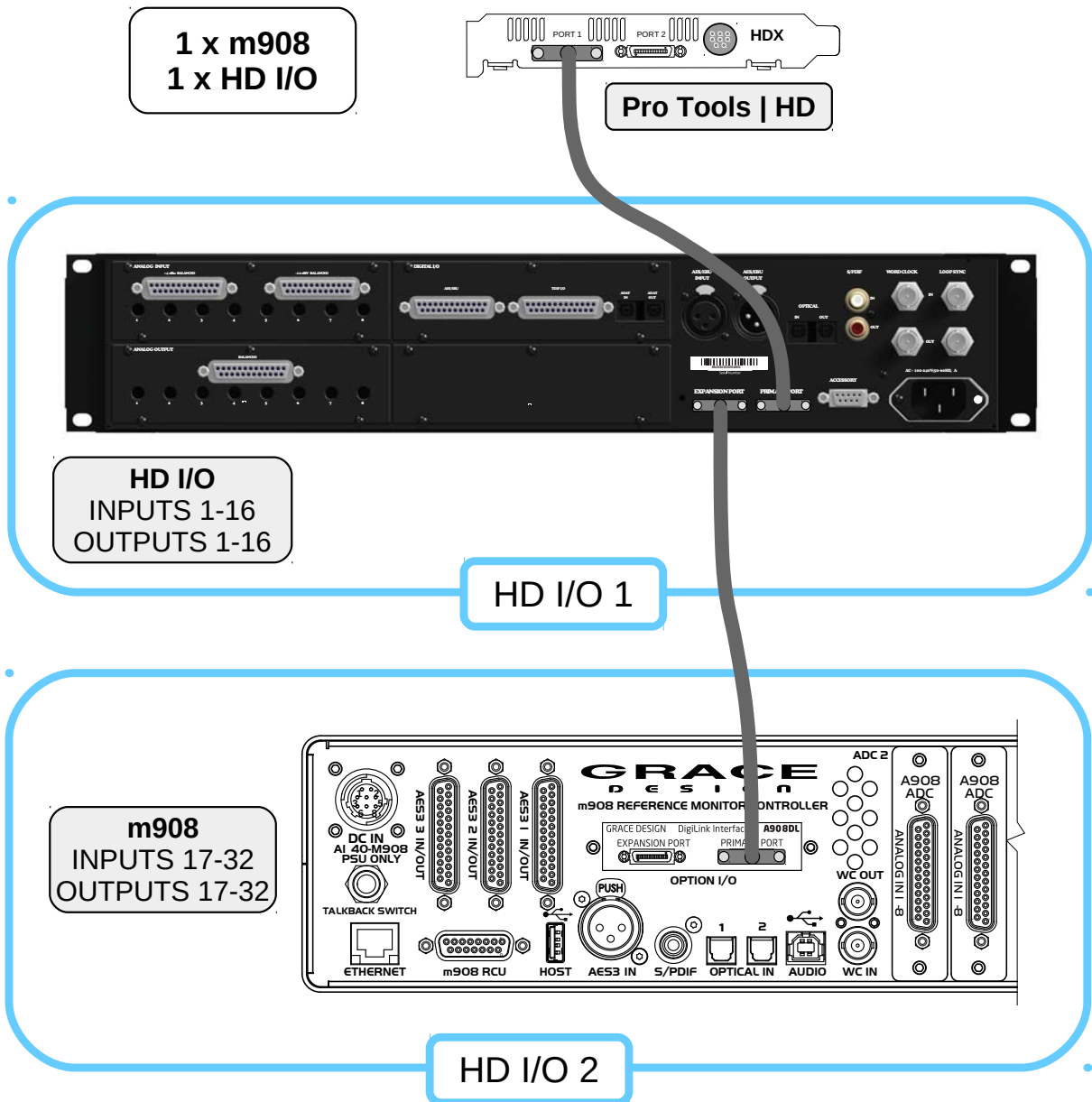
Illustration 106: 16 Channel Digilink Connections

When the m908 is connected to an HDX or Native port along with another interface the m908 must be second in the chain. This is because the m908 emulates two HD I/O devices and the EXPANSION port is not active.

When the m908 is connected in the second position it will show up in Pro|Tools Hardware

Setup as HD I/O #2 and receive audio from Pro|Tools on the first 16 channels of the m908 Digilink interface.

7.5 Phono Preamp Option Module

The m908 phono module is a high performance RIAA preamplifier. This option is available on new and existing units. Please contact your Grace Design dealer or distributor for details.

The phone module requires that the ADC1 option is installed as it uses the Unbalanced analog inputs. To select the RIAA phone preamp the Unbalanced Input Mode must be set for phono. See System Setup for details.

The phono amplifier can be optimized for a wide range of phono cartridges via a number of internal jumper settings. Moving magnet cartridges can be accommodated with the 40dB gain setting and 47k Ω input impedance. Low output moving coil cartridges can be accommodated with the 60dB gain setting and 100 Ω input impedance. The phono module also provides variable input loading from 0-500 Ω with VR1 and VR2. Connect an Ohm meter to the unbalanced input jacks to set the input loading to the cartridge recommended setting.

To access these jumpers, remove the top cover of the m908 and refer to the Jumper Location diagram below. Be sure to unplug the AC power cord before removing the m908 top cover.
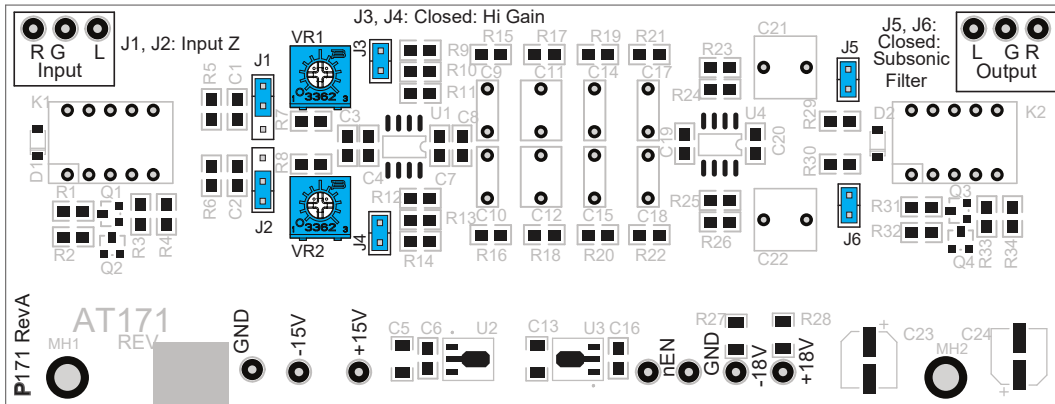
PHONO INPUT MODULE PCB
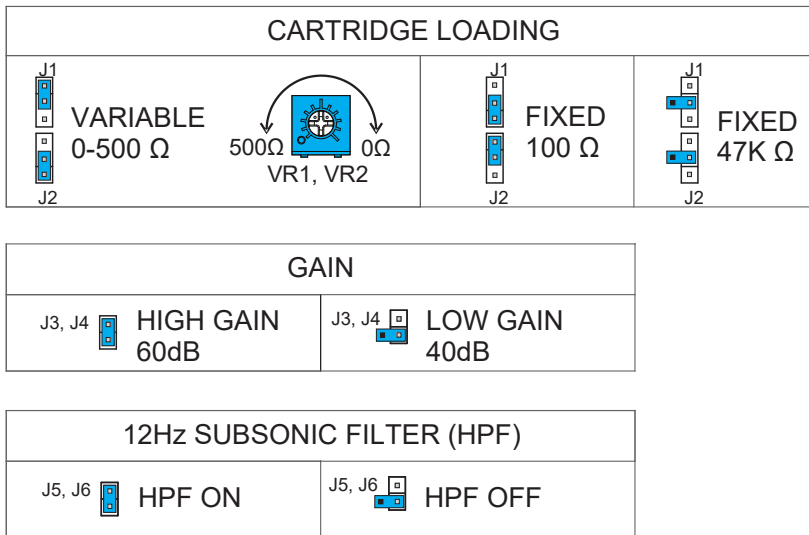
Illustration 107: RIAA Phono Module Jumper Locations

8 Web/Network Setup

8.1 m908 Network Settings

The m908 features a web based user interface that can be used for configuring all of the Setup functions that are available in the setup mode on the RCU. To access the m908 web UI from a computer the m908 must be connected to a network.

The m908 comes from the factory configured to receive an IP address automatically from a

DHCP server. It can be configured to use a static IP address as well. Below is a list of the steps to configure the m908 for web access. Refer to the System Setup section of this manual for details making changes.

8.1.1   Network Configuration

1. Set the network IP address for your network type. Most networks will have a DHCP server to assign an IP address automatically to the m908. If your network does not have a DHCP server then set the **tcp/ip mode** to <static>. Then set the **ip address** to the desired number. Next set the **subnet mask** according to your requirements. Finally, exit setup to save the changes and wait 5 seconds for the file write to complete. Then reboot the m908 for the network settings to take effect.

2. Set the password. This password will be required to log in to the web UI. Note that the password can be changed from the web UI.

8.1.2   Web UI Access

1. To access the web UI enter either of the following in to the address field of your browser:

2. http://m908-XXXX.local where "XXXX" is the last 4 digits of the m908 serial number. (this can be found on the bottom panel of the RCU) The ".local" assures that the computer will look for the m908 in a local DNS.

3. http://xxx.xxx.xxx.xxx where "xxx.xxx.xxx.xxx" is the ip address of the m908 as displayed in the Setup>System>ip address field.

4. The m908 will respond with a password window. Enter the m908 password and check the "Auto Login" check box to have the web UI remember the password for future sessions. The default password is "grace". Click the Connect button to open the m908 home page.

9   USB Audio Setup

9.1 Computer Audio Requirements

The m908 ships with a standard USB type A to type B cable. The type A connector plugs in to the computer and the type B connector to the m908 USB Audio jack.

The m908's asynchronous mode USB controller supports standard driverless operation on Mac OSX since version 10.8.5. Sample rates up to 192kHz are supported.

For Windows 7, 8, 8.1, and 10 PC computers, users will need to download and install the Grace Design m908 USB ASIO audio driver. This driver can be found on our website:

http://www.gracedesign.com/support/support.htm

For Windows 7 make sure that your computer has the latest updates from Microsoft or driver installation may fail.

Installation and operation instructions can be found in the following chapter: <u>Windows USB ASIO Driver</u>.

On Mac computers driver-less operation basically means the m908 will automatically show up in your computer's list of supported audio devices as 'm908'. The m908 will provide 8 input channels and 24 output channels. In most cases, simply choose the m908 as your audio recording or playback device and the system will work.

In the event that you have any setup issues which aren't addressed by these resources, feel free to contact us directly at 303.823.8100, Monday through Friday 9 – 5, MST.

9.2 Windows USB ASIO Driver

9.2.1 Installing USB ASIO Driver

The m908 USB ASIO driver is necessary for operating the m908 as an I/O device on a Windows computer.

1. Connect the m908 to you computer.
2. Download the installer .zip package from our website: http://www.gracedesign.com/support/support.htm Extract the .zip file, and run the installer .exe file."GraceDesign_TUSBAudio_v4.0.0_2017-04-24_setup.exe"
3. Follow the steps illustrated below (It may take a few moments for the user prompts to appear once the program is running).
4. Depending on the version of Windows your computer is running you may see a security window similar to one of the images shown above. When prompted whether or not you want to allow the program to make changes or if you want to run this file - click "Yes" or "Run" to continue.:
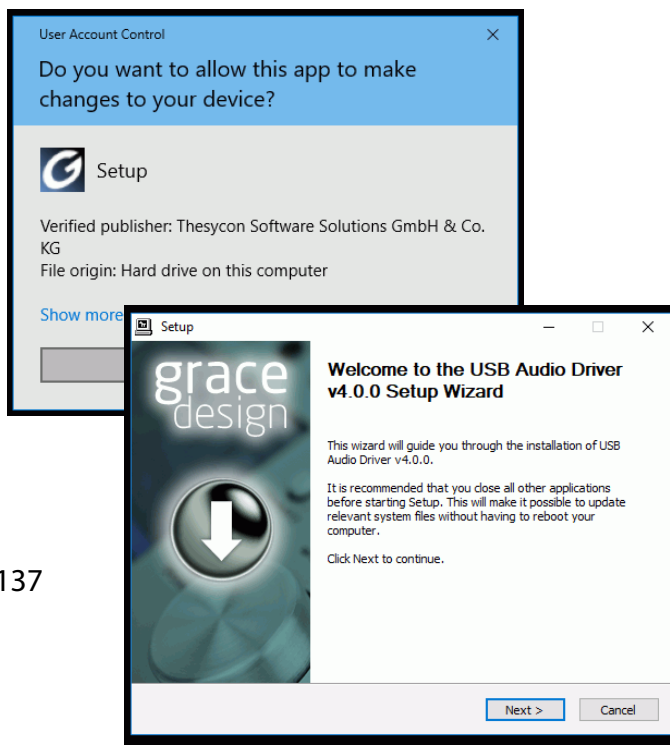
5. The driver installer will run and display the following dialog. Click "NEXT" to start the installation.

6. Choose the location you wish install the driver and click "Install" to proceed. It is recommended to use the default installation folder.
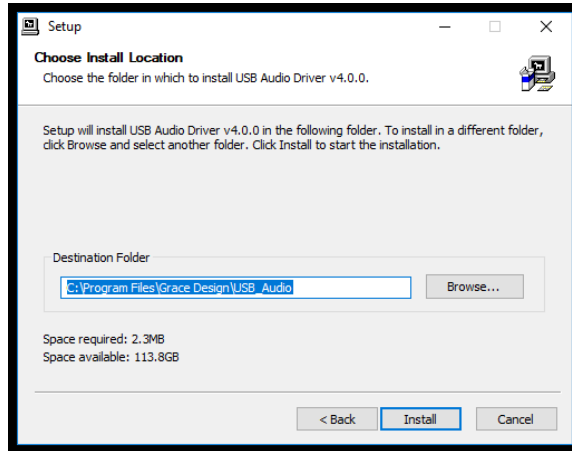
7. When the installation is complete click "NEXT"
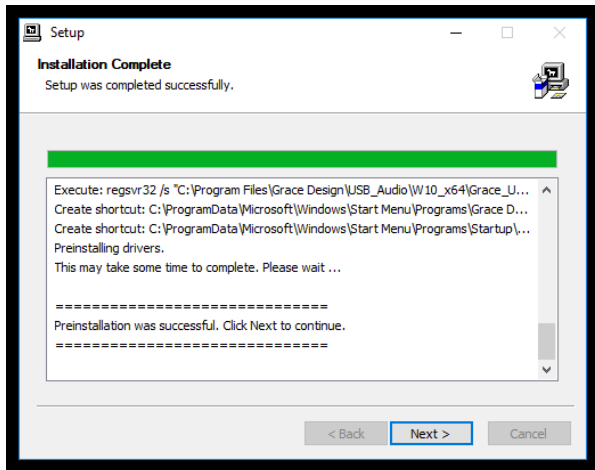
8. To exit the installer click "FINISH"

9.2.2   Using The USB ASIO Driver

1. With the driver installed properly and the m908 connected to your computer, there will be a blue and white "G" icon in the system tray, or "Grace Design USB Audio Control Panel" in the start menu. Simply click this icon to open the driver control panel. There are three tabs in this panel. The Status panel will be visible initially. The Status panel shows the connected device and the currently defined OS sample rate:
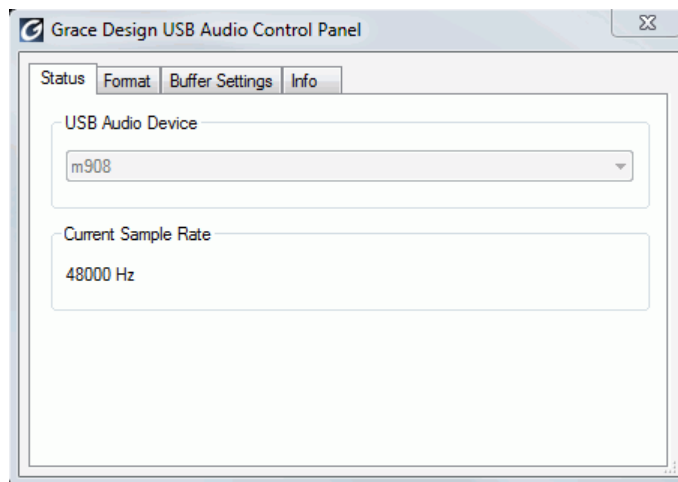
Illustration 108: USB Audio Control Panel Status

2. Buffer Settings displays the USB Streaming mode and the ASIO Buffer size:
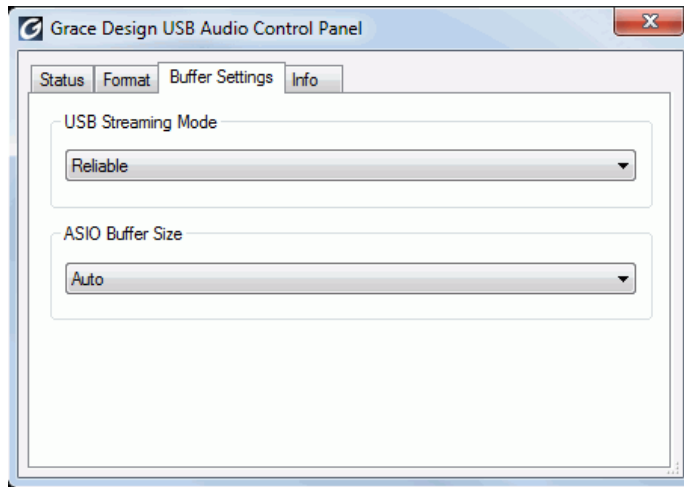
Illustration 109: USB Audio Control Panel Buffer Settings

3. The USB Streaming Mode sets the computer's USB latency performance, ranging from very low to high latency. The lower the latency, the greater the load will be on your computer's resources, which may result in errors or distortion. The ASIO Buffer Size also affects latency, the smaller the buffer the lower the latency. This setting is interactive with the Streaming Mode. The panel will display a warning if the buffer size is too small for the selected streaming mode and sample rate. You will need to experiment with these options to find a suitable setting for your computer.

4. The USB Format tab contains the Input and Output formatting options for the m908. The number of channels that can be streamed from the computer host is dependent on the maximum sample rate required.
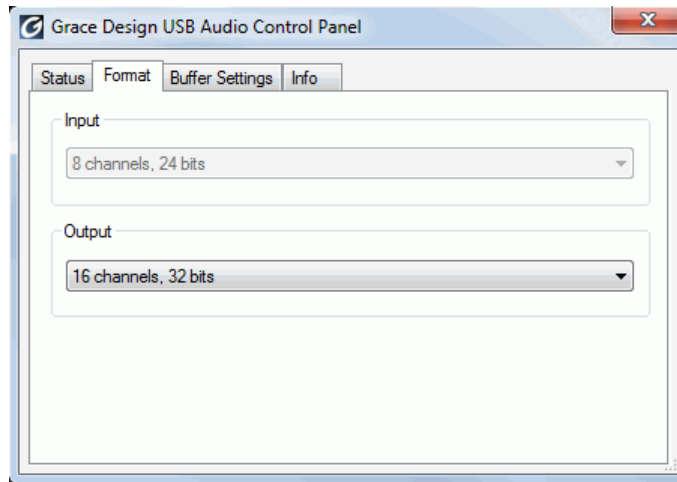
Illustration 110: USB Audio Control Panel Format

The m908 USB Audio Interface supports the following sample rate/channel counts. Note that the input channels are currently not routable in the m908.

USB Channel Count		
Output Format	Sample Rates	Max Ouptut Channels
24 channels, 32 bits	44.1-48kHz	24
16 channels, 32 bits	44.1-96kHz	16
8 channels, 32 bits	44.1-192kHz	8

5. Info displays the Device and the Driver information.
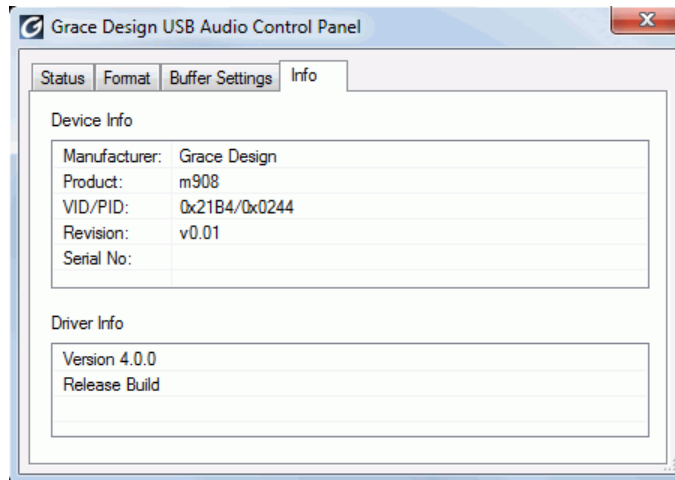
Illustration 111: USB Audio Control Panel Info

10   Firmware Upgrades

The m908 is designed so that system firmware upgrades are simple. The current m908 firmware is available on our website:

https://gracedesign.com/support-documents/

Upgrading firmware from 1.0.x to 2.1.x requires a migration procedure which replaces the entire operating system of the m908. Once the system migration is complete further updates will follow the same update procedure.

10.1.1   Firmware Migration from 1.0.x to 2.1.x

The migration from 1.0.x to 2.1.x involves completely replacing the ACU processor operating system with a new linux distribution. To do this the m908 needs to be able to boot from an external USB drive connected to the ACU USB Host connector. You will need to create a linux formatted USB drive from the disk image downloaded from the Grace Design website support page. Also, disk image burning software is required for this. Refer to the Grace Design website support page for the migration package and detailed instructions.

10.1.2   Firmware Update Procedure

The procedure for upgrading firmware from 1.x.x > 1.x.x or 2.x.x > 2.x.x is on the Grace Design website support page. Refer to the upgrade package for upgrade instructions.

1.

11 Electrical Specifications

See document m908 System Specifications which can be found at:

https://gracedesign.com/support-documents/

Grace Design reserves the right to update or change the electrical specifications at any time.

12 Block Diagrams

See m908 System Flow Diagram.

13   Cable and Connector Diagrams

DB25 Digital AES3 (Tascam Pinout)				
CH#	DB25 Pin#	AES3 1 IN/OUT	AES3 2 IN/OUT	AES3 3 IN/OUT
	13	GPIO 1	GPIO 2	GPIO 3
IN 1-2	25	GND	GND	GND
	24	Positive(+)	Positive(+)	Positive(+)
	12	Negative(-)	Negative(-)	Negative(-)
IN 3-4	11	GND	GND	GND
	10	Positive(+)	Positive(+)	Positive(+)
	23	Negative(-)	Negative(-)	Negative(-)
IN 5-6	22	GND	GND	GND
	21	Positive(+)	Positive(+)	Positive(+)
	9	Negative(-)	Negative(-)	Negative(-)
IN 7-8	8	GND	GND	GND
	7	Positive(+)	Positive(+)	Positive(+)
	20	Negative(-)	Negative(-)	Negative(-)
OUT 1-2	19	GND	GND	GND
	18	Positive(+)	Positive(+)	Positive(+)
	6	Negative(-)	Negative(-)	Negative(-)
OUT 3-4	5	GND	GND	GND
	4	Positive(+)	Positive(+)	Positive(+)
	17	Negative(-)	Negative(-)	Negative(-)
OUT 5-6	16	GND	GND	GND
	15	Positive(+)	Positive(+)	Positive(+)
	3	Negative(-)	Negative(-)	Negative(-)
OUT 7-8	2	GND	GND	GND
	1	Positive(+)	Positive(+)	Positive(+)
	14	Negative(-)	Negative(-)	Negative(-)

DB25 Analog (Tascam Pinout)				
CH#	DB25 Pin#	CR Out 1, CR Out 2	CUE/CR EXT.	ADC In (option)
	13	Not Connected	Talkback Out +3.3V	Not Connected
1	25	GND	GND	GND
	24	Positive(+)	Positive(+)	Positive(+)
	12	Negative(-)	Negative(-)	Negative(-)
2	11	GND	GND	GND
	10	Positive(+)	Positive(+)	Positive(+)
	23	Negative(-)	Negative(-)	Negative(-)
3	22	GND	GND	GND
	21	Positive(+)	Positive(+)	Positive(+)
	9	Negative(-)	Negative(-)	Negative(-)
4	8	GND	GND	GND
	7	Positive(+)	Positive(+)	Positive(+)
	20	Negative(-)	Negative(-)	Negative(-)
5	19	GND	GND	GND
	18	Positive(+)	Positive(+)	Positive(+)
	6	Negative(-)	Negative(-)	Negative(-)
6	5	GND	GND	GND
	4	Positive(+)	Positive(+)	Positive(+)

	17	Negative(-)	Negative(-)	Negative(-)
7	16	GND	GND	GND
	15	Positive(+)	Positive(+)	Positive(+)
	3	Negative(-)	Negative(-)	Negative(-)
8	2	GND	GND	GND
	1	Positive(+)	Positive(+)	Positive(+)
	14	Negative(-)	Negative(-)	Negative(-)
Note: For unbalanced output operation leave the Negative (-) pin open				

DB15 RCU and ACU Connectors		
PIN#	ACU Function	RCU Function
1	RX-	RX-
2	GND	GND
3	TX-	TX-
4	+5VDC	NC
5	GPIO	NC
6	TALKBACK- IN	TALKBACK- OUT
7	HEADPHONE LEFT OUT	HEADPHONE LEFT IN
8	HEADPHONE GND	HEADPHONE GND
9	RX+	RX+
10	TX+	TX+
11	+6.5VDC OUT	+6.5VDC IN
12	+3.3VDC	NC
13	TALKBACK+ IN	TALKBACK+ OUT
14	GND	GND
15	HEADPHONE RIGHT OUT	HEADPHONE RIGHT IN
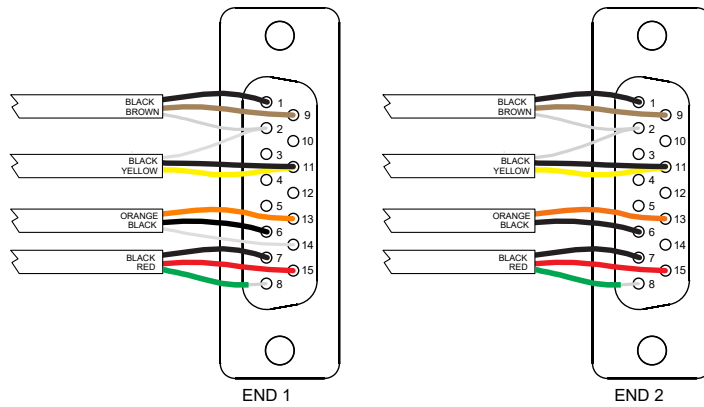
Illustration 112: RCU Cable Termination
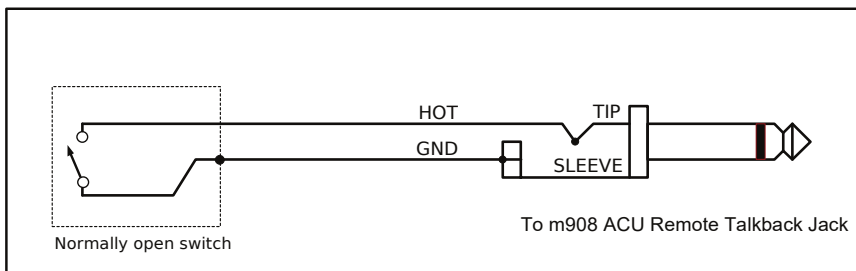
Illustration 113: Remote Talkback Switch Wiring
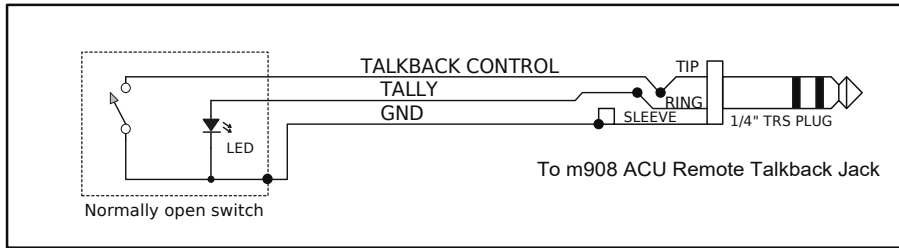
Illustration 114: Remote Talkback Switch with LED Tally
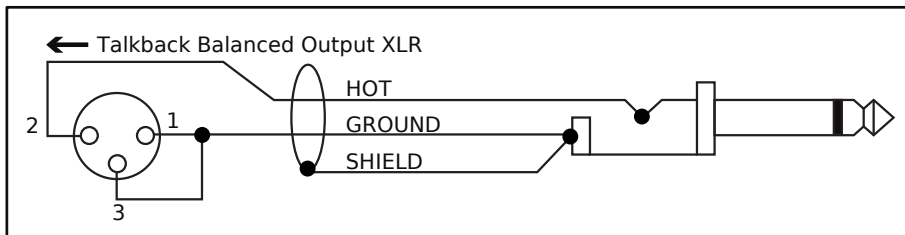
Illustration 115: Talkback Out Unbalanced Wiring

14 Dimensions

ACU, RCU, and PSU dimension can be found in the m908 System Specifications document which can be found at: https://gracedesign.com/support-documents/.

Warranty Information

- Grace Design warrants this product to be free of defective parts and workmanship for a period of five years. This warranty period begins at the original date of purchase and is transferable to any person who may subsequently purchase the product during this time.
- This warranty excludes the following conditions: normal wear and tear, misuse, customer negligence, accidental damage, unauthorized repair or modification, cosmetic damage and damage incurred during shipment.
- During the time of this warranty, Grace Design will repair or replace, at its option, any defective parts or repair defective workmanship without charge, provided the customer has appropriate proof of purchase and that the product has its original factory serial number.
- In order for Grace Design to provide efficient and timely warranty service, it is important that you mail the completed warranty registration card enclosed with all of our products within 10

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- In no event will Grace Design be liable for lost profits or any other incidental, consequential or Exemplary damages, even if Grace Design is aware of the possibility of such damages. In no event will Grace Design's liability exceed the purchase price of the product.
- This warranty gives the customer specific legal rights. The customer may also have other rights, which vary from state to state. Some states do not allow limitations on implied warranties or consequential damages, so some of the limitations of the above may not apply to a particular customer.

15   Illustrations Index

16 Manual Revision History

Manual Revision History			
Rev.	**Description**	**Date**	**Initials**
A	Initial Release	9/30/2019	MBG
B	Changed L-R Setup parameter information. Edit Firmware update procedure Update DB15 Connector Pin function table Updated the Workflow Template table	10/9/2019	MBG
C	Update RCU cable wiring diagram to show no GND connection on pin 14. Update ACU and RCU DB15 pin function table. Update Ext Talkback gain range. Update Talkback XLR Output Connector description, Corrected references to unity gain in 5.3.4 and 6.3.14 Replace Room EQ screenshot in Setup>CR1, CR3, CR3 Update Digilink module option section.	10/11/2019	MBG
D	Update Room EQ Q range specifications Update User ABC Switch Setup to show downmix options. Update Setup CR Outputs to explain Headphone CUE/CR EXT routing conflicts. Change Cue Unity Gain setting	10/26/2019	MBG
E	Update max power consumption specs Update CR1, CR2, CR3 Setup routing conflicts on CUE/CR EXT with Headphones enabled. Update Clock Status table to include PLL Lock status Updated Level Indicator thresholds.	11/13/2019	MBG
F	Add Direct Input Offset Mode details in section 5 Add GPIO Trigger mode to GPIO table. Add GPIO Logic definitions to GPIO setup. Add warnings to not turn of power when workflows are being saved. Fix CR Output hardware conflict table	11/26/2019	MBG
G	Update Meter Routing flow diagram. Update Direct Access Input Offset saving information. Add Setup Input speaker icon color information.	12/10/2019	MBG
H	Add ADC input routing drawing. Add Control Room Output routing conflict information.	1/30/2020	MBG

I	Change Meter routing signal source options and flow diagram to match firmware. Changed RCU and EXT Talkback switch mic assign setup options to match firmware capabilities.	2/22/2020	MBG
J	Add Import Workflow from USB details Add new Control Room EQ setup information Update Audio Processing flow diagram Illustration 2. Added meter over-hold mode description to Setup>System	4/8/2020	MBG
K	Add CUE input mute functions Add Talkback Activation Mode settings Add Talkback output signal options Add GPIO enhancements Add ABC Bass Management Bypass function	10/13/2020	MBG
L	Add Ravenna/AES67 Option Module Info	3/25/2021	MBG
M	Add new sub modes and update bass management flow diagram Add LFE filter slope parameters Add Unbalanced In mode with RIAA phono option.	9/3/2021	MBG
N	Removed references to MADI option module	2/2/2022	MBG
O	Add web GUI configuration info. Synchronized all setup parameter ranges, step sizes, and list order with RCU and web UI. Removed references to remaining room EQ bands.	8/10/2022	MBG
P	Fixed errors in Web UI access 8.1.2, removed link speed menu item.	3/29/2023	MBG