



AERO.qc

**Audio Quality Controller with
UPMAX II, Loudness Metering, Monitoring,
HD/SD-SDI, and Optional Dolby Decoding**

Preliminary User Guide





AERO.qc

**Audio Quality Controller with
UPMAX II, Loudness Metering, Monitoring,
HD/SD-SDI, and Optional Dolby Decoding**

Preliminary User Guide

Release Date: June, 2012
Software Version: 2.20.29 and later



Linear Acoustic Inc.
Phone: 717-735-3611
FAX: 717-735-3612
Web: www.LinearAcoustic.com

Warranty Information

DISCLAIMER OF WARRANTIES: Products manufactured by Linear Acoustic are warranted against defects in material and workmanship for a period of two (2) years from the date of purchase. **THERE ARE NO OTHER IMPLIED OR EXPRESS WARRANTIES AND NO WARRANTY FOR MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE.**

During the warranty period Linear Acoustic Inc. will repair, or at our discretion replace, components which prove to be defective, provided the unit is returned, shipped pre-paid to us directly with a return authorization (RA) number clearly marked on the packaging. **Please note, this RA number must be present or package will be refused and returned to sender.**

All requests for repairs MUST include the unit serial number to ensure quick and accurate service.

DEFECTS CAUSED BY UNAUTHORIZED MODIFICATIONS, MISUSE OR ACCIDENTS, UNAUTHORIZED CUSTOMER REPAIRS, OR ANY FURTHER DAMAGE CAUSED BY INADEQUATE PACKAGING FOR SERVICE RETURN ARE NOT COVERED BY THIS WARRANTY.

PLEASE SAVE THE SHIPPING CARTON AND ALL PACKING MATERIALS. FAILURE TO RETURN UNIT IN ORIGINAL SHIPPING CARTON AND PACKING MATERIALS WILL RESULT IN A CHARGE FOR NEW SHIPPING MATERIALS.

LIMITATION OF PERIOD OF ACTION ON CONTRACT: No action, regardless of form, arising out of the transactions under this agreement may be brought by buyer, its successors, agents and/or assigns, more than three years from date of purchase.

LIMITATION OF LIABILITY: It is understood and agreed that Linear Acoustic's liability whether in contract, in tort, under any warranty, in negligence or otherwise shall not exceed the cost of repair or replacement of the defective components and under no circumstances shall Linear Acoustic be liable for incidental, special, direct, indirect or consequential damages, or loss of use, revenue or profit even if Linear Acoustic or its agents have been advised, orally or in writing, of the possibility of such damages.

This product contains Audyne dynamic range and loudness control processing and is manufactured under license.

This product optionally contains Dolby Digital (AC-3)/Dolby E decoding and is manufactured under license from Dolby Laboratories, Inc. The decoder contains confidential unpublished works and is protected by US and foreign patents and is copyright 2007 Dolby Laboratories, Inc.

Linear Acoustic, the "LA" symbol, AERO.qc, AEROMAX, AutoNorm, MetaMAX, AutoMAX-II, AutoVoiceover, AutoVOX, CrowdControl, UPMAX, and UPMAX-II are trademarks of Linear Acoustic Inc., all other trademarks remain the property of their respective owners.

Table of Contents

List of Figures	vii
Chapter 1: Introduction	5
1.1 Principles of Operation.....	5
1.2 Reference Levels and Line Up Features.....	6
1.3 Powering Up and Shutting Down.....	6
1.4 Reset and Upgrades.....	7
1.5 Warranty and Feedback.....	7
Chapter 2: Connections and Quick Setup	9
2.1 Unpacking and Inspection.....	9
2.2 Installation.....	9
2.3 Rear Panel.....	10
2.3.1 Connection Ports.....	10
2.4 Quick Setup Notes.....	12
2.5 Password Programming.....	12
Chapter 3: Applications	13
Chapter 4: Detailed Operation	19
4.1 Menu Navigation.....	19
4.1.1 AERO.qc.....	19
4.1.2 System Settings.....	26
Chapter 5: Troubleshooting	29
5.1 Problems and Possible Causes.....	29
5.1.1 Unit won't power on.....	29
5.1.2 The "LA" symbol is flashing red.....	29
5.1.3 Output Audio Clicks and Pops.....	30
5.1.4 Audio Sounds Strange.....	30

5.1.5	Audio Issues Disappear In Bypass	30
5.1.6	Audio Fixed by Re-boot.....	31
5.1.7	Audio Has Artifacts	31
Chapter 6: Monitor Setup		33
6.1	Monitor Calibration (Analog Outputs)	33
6.1.1	Main Channel SPL Adjustment.....	33
6.1.2	Subwoofer SPL Adjustment	34
6.1.3	Speaker Equalization.....	35
6.1.4	Speaker Delay.....	35
6.1.5	Listen!.....	35
Chapter 7: Presets & Processing Adjustment		37
7.1	Description of Factory Presets & Adjustments	37
7.1.1	Processing Structure.....	38
7.1.2	Adjusting Processing.....	39
Chapter 8: Specifications		47

List of Figures

1-1	Audio Block Diagram	6
2-1	Rear Panel	10
3-1	AERO.qc Used for ingest of discrete audio channels	14
3-2	AERO.qc Used for ingest of SDI embedded audio channels	15
3-3	AERO.qc used for decode of Dolby E audio and loudness measurement during ingest	16
3-4	AERO.qc used for live production	17
4-1	Locked Main Screen (left); Unlocked Main Screen (right)	19
4-2	Monitor Source Select	26
7-1	General signal flow of the AERO.qc processing core	39
7-2	Dynamic Range Control Menu	39

Chapter 1: Introduction

The Linear Acoustic AERO.qc Audio Quality Controller is an all-in-one unit that combines ITU-R BS.1770 loudness measurement with tools to allow manual or automatic audio management. It also has extensive monitoring functions including speaker level, equalization and delay calibration and downmix emulation.

The unit facilitates the quick creation or ingest of content to a known standard target. Loudness and surroundfield are both kept consistent and results are displayed graphically so that no interpretation is required better ensuring that all operators will have the same high quality result.

The AERO.qc provides the following features:

- Triple ITU-R BS.1770 loudness meters (fast, medium and slow) with numeric and logical color graphic indication of results: green=good
- True peak indication
- Selectable and adjustable upmixing of Main Input 1/2 (Left/Right front) via industry standard UPMAX and UPMAX-II stereo to 5.1 channel upconversion algorithms
- LtRt or LoRo downmix available full-time
- Monitor section including speaker calibration and downmix emulation
- Redundant power supplies
- Loudness and dynamic range control for up to six channels of audio; includes input AGC, multiband AGC and limiting, distortion-free look-ahead peak limiting, and selectable multiband source noise reduction
- HD/SD-SDI I/O for embedded audio
- Optional built-in dual Dolby Digital (AC-3)/Dolby E decoding

1.1 Principles of Operation

The AERO.qc processing structures support up to six channels of main audio input. Discrete 5.1 channel audio can be applied, or two channel audio can be upmixed to 5.1 channels if desired.

Figure 1-1 shows the internal audio path of the AERO.qc. It should be noted that this diagram is a general representation of signal flow. Please consult the appropriate section of this manual for specific functionality.

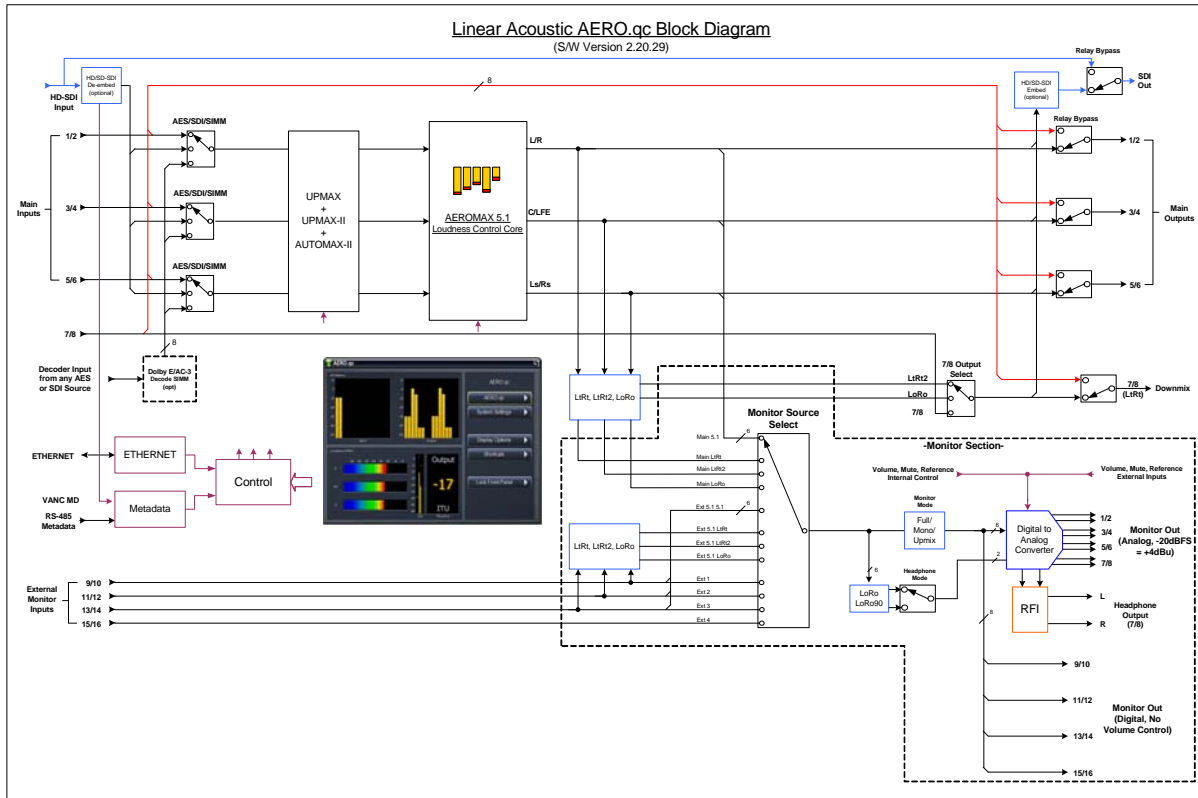


Figure 1-1 Audio Block Diagram

1.2 Reference Levels and Line Up Features

The AERO.qc is designed to support a standard reference level of -20dBFS via its digital inputs. Other levels can be supported as the slow-moving Input AGC will easily compensate for any level differences. Please see Chapter 4 for detailed information.

1.3 Powering Up and Shutting Down

The AERO.qc automatically power on when connected to AC power.

NOTE: It is strongly recommended that both power supplies are used at all times; best practice is to apply AC power from alternate sources via uninterruptable power supply (UPS) units.

The Linear Acoustic logo blue LEDs will illuminate first and the unit will then boot and the bypass relays will switch the unit into the audio path.

NOTE: *It may take up to 90 seconds for the boot and configuration loading process to complete. During this time, clocks will be switching and may require several additional seconds to stabilize.*

Shutting the unit off is as simple as removing AC power.

If there is a need to bypass the unit, the best way is via Master Bypass in the System Settings menu.

1.4 Reset and Upgrades

To reset the unit, simply follow the power off procedure above, then power back on.

For software updates, the factory will provide a USB updater with instructions on how to update software.

WARNING: *Do not connect any devices to the USB connector on the rear panel without specific instructions from the factory!*

1.5 Warranty and Feedback

Please take a moment to fill out and return the warranty card included with the unit and drop it in the mail. This will enable us to contact you if there are any software or documentation issues. Also, we are very interested in your feedback. This unit was designed based on input gathered from many broadcast engineers and it will evolve further thanks to on-going suggestions and comments from users. We look forward to hearing from you.

Chapter 2: Connections and Quick Setup

This chapter covers all required connections for the AERO.qc.

2.1 Unpacking and Inspection

Before unpacking the unit, inspect the outer carton for shipping damage. If the carton shows damage, inspect the unit in those areas. Please save the carefully designed shipping carton and packing materials. In the unlikely event that the unit needs to be returned to the factory, alternate cartons or packing materials may not be adequate and can cause damage not covered by warranty.

The following essential items are provided with the unit:

- Bag containing:
 - One or two IEC power cords (style matches country of order);
 - this manual, and a small piece of SWAG
- Warranty information: Please fill out and return the warranty card to Linear Acoustic to ensure your software and documentation are kept up to date.

2.2 Installation

AERO.qc installation requires:

- two standard rack space units with ADEQUATE VENTILATION (the unit vents from the front side slots);
- standard 75-Ohm BNC cables for digital signal connections;

To connect to digital equipment with 110-Ohm XLR connectors, use impedance-matching transformers (available from Canare, Neutrik and other manufacturers).

- Proper selection of reference. The unit will default to internal 48kHz, but for proper timing should be set to AES Ref In, AES 1, or SDI.

2.3 Rear Panel

The rear panel of the AERO.qc contains its electrical I/O.

2.3.1 Connection Ports

All of the AERO.qc's connections are on the rear panel and are described in detail below. See Chapter 5: *Specifications* for specific pinouts.

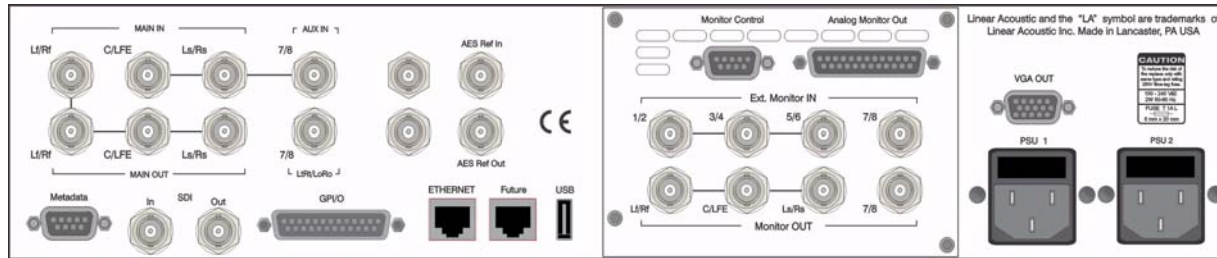


Figure 2-1 Rear Panel

- **Main In:** Connect the 48kHz PCM signals to these inputs. The input channels are arranged as follows: 1/2 = Left front/Right front, 3/4 = Center/LFE, 5/6 = Left surround/Right surround. 7/8 is currently not used.
- **Aux A In:** Currently not used.
- **AES REF:** Connect a 48kHz AES plant reference signal to this connector (**not video or wordclock**). The signal may be an AES Black or AES signal containing audio. All Main Audio Inputs must be synchronous and locked to the AES reference signal applied to the AERO.qc.

NOTE: *Appropriate reference must be selected for proper operation.*

- **Ext. Monitor IN (OPTION):** Four inputs allowing monitoring of 5.1 + 2 external sources or four external stereo sources.
- **Main Audio Output:** Main 48kHz processed digital audio outputs with same channel configuration as the inputs on 1-6. The output of the extra stereo +2 program is available on 7/8 or a compatible LtRt downmix of the main program can be selected.
- **Aux A Out:** Currently not used.
- **Aux B Out:** Currently not used.
- **Monitor OUT:** Digital monitor outputs, pre volume and mute.

-
- **Monitor Control:** Inputs for an external rotary encoder for volume plus Mute and Return to Reference switches and LED indicators. See Chapter 8 for pinout.
 - **Analog Monitor Out:** +4dBu output controlled by front panel or external volume control and mute.

NOTE: *Analog outputs are normalized to produce +4dBu with a -20dBFS input signal; users must provide proper gain matching for best results.*

- **Metadata I/O:** RS-485 connection accepts the metadata output of any Dolby equipment (DP572, DP570, etc...) or any Dolby-compatible metadata source. Used to control upmixing and other functions.
- **SDI Input:** Any of the 16 possible audio channel from an applied HD/SD-SDI signal can be de-embedded and used in place of or along with AES inputs. This can be selected in the System Options->I/O Options menu on a pair-by-pair basis.
- **SDI Output:** Up to 16 audio channels, including any of the Main outputs. Audio channels present in the applied SDI signal can also be passed through in a bit-accurate manner, allowing for example pre-encoded Dolby E or AC-3 bitstreams to pass untouched. AES outputs remain active in parallel with embedded audio.
- **GPI/O:** Used to control upmixing and other functions.
- **ETHERNET:** Automatically senses 100 or 1000BASE-T Ethernet connectivity and is used for connection to an available remote control application.
- **Future:** Extra RJ-45 connector for future I/O use.
- **USB:** Used for software updates via factory instruction. Please Do Not connect any USB devices to this connector without factory guidance as this may affect the performance of the unit.

NOTE: *Please do not connect anything to the USB connector without specific instructions from the factory. Permanent damage can occur!*

2.4 Quick Setup Notes

The Linear Acoustic AERO.qc is configured at the factory and is ready to go on the air after making the proper input and output connections and determining if upmixing is desired.

- Select appropriate reference from the System Options->I/O Options menu (AES Ref, AES 1, SDI, or Internal 48kHz) and make sure the selected reference is applied to the unit. If reference is missing, the “double L” signal in the upper right corner of the screen will flash red. For proper operation it should not be flashing.
- Apply audio sources to Main AES inputs and/or SDI Input.
- On a pair-by-pair basis, select the inputs to the processor in the System Options->I/O Options menu.
- You should see input audio on the top left set of front panel meters, and output audio on the top right set of front panel meters.
- You should now have audio outputs on the Main AES outputs, as well as the Monitor Outputs.
- Input audio can be monitored by entering Full Screen mode (press the Right side of the Menu navigation ring).
- If the loudness is incorrect, it can be adjusted by toggling down to the “Input Attenuator” screen:
 - Press Down on the Menu navigation ring while still in the full screen mode until “Input Attenuator: xdB” is shown at the top of the screen
 - Rotate the Menu knob to adjust the level until the meter displays a good result

The best way to learn the unit is to explore the different settings with audio applied and monitored. There are many factory presets to support many different tastes. Some presets may not sound much different at first, but will perform very differently depending on program content.

2.5 Password Programming

After setup, users may wish to password-protect the AERO.qc. This will prevent unauthorized or accidental changes to the settings, however please do not forget the password!

Chapter 3: Applications

The Linear Acoustic AERO.qc Audio Quality Controller was designed to provide all of the tools necessary to produce, ingest, and correct high quality stereo and multichannel audio content:

- Four ITU-R BS.1770 meters that simultaneously measure audio using different integration times for faster and more accurate judgment of short-term and long-term loudness.
- Stereo to 5.1 channel upmixing using industry standard UPMAX or UPMAX-II algorithms
- Fully-featured monitoring section that provides speaker equalization and delay calibration and a real-time spectrum analyzer
- Selectable Multiband, multistage automatic loudness and dynamic range control
- HD/SD-SDI I/O and dual power supplies
- Optional Dolby Digital/Dolby E decoding (auto-sensing)

The applications for this device are many. Below a few simple scenarios are shown in block diagram form to highlight some typical use cases. Specific applications can usually be derived from one of these basic setups.

AERO.qc In Baseband Ingest Application

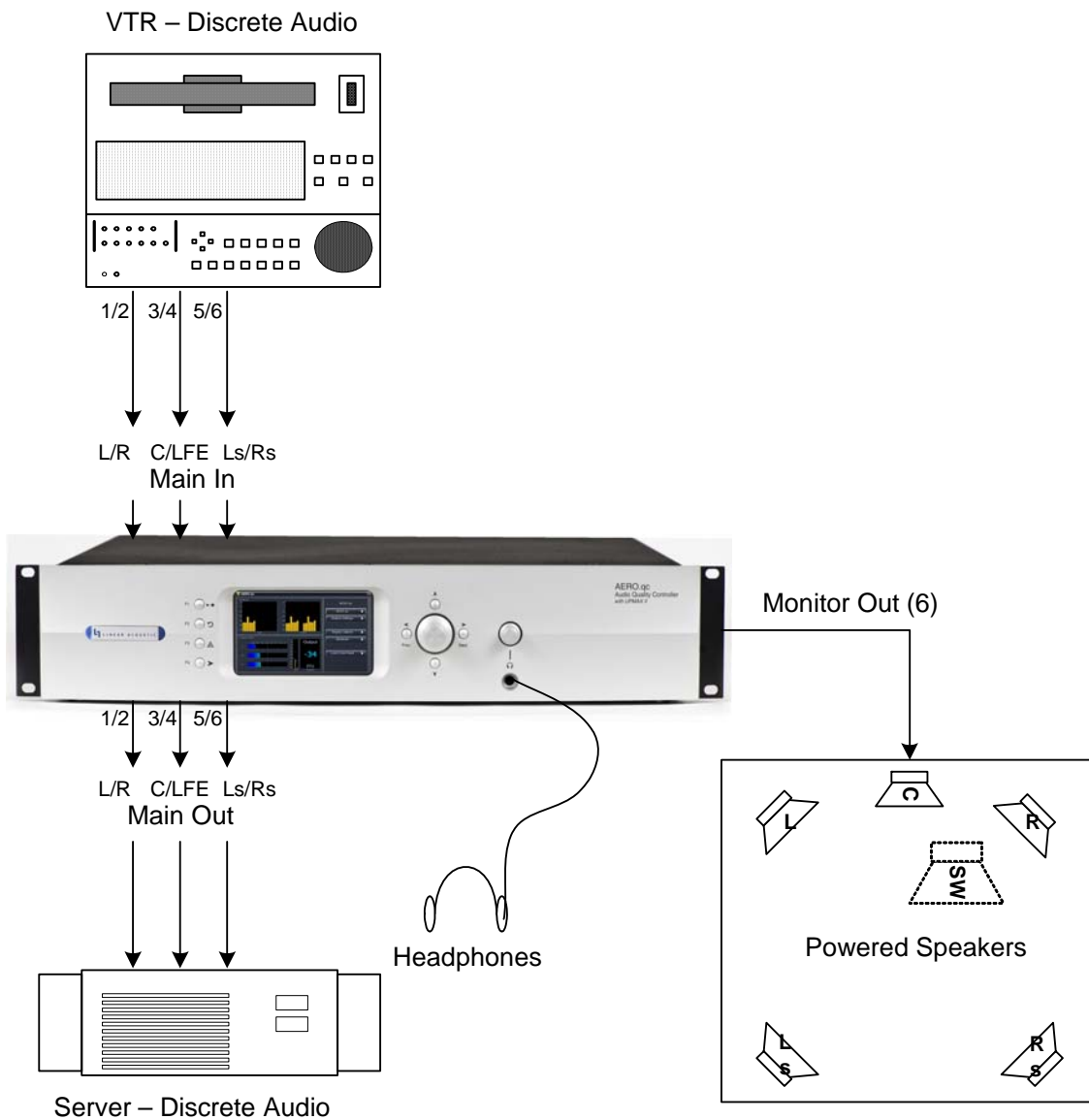


Figure 3-1 AERO.qc Used for ingest of discrete audio channels

As shown above, discrete audio can be played back from tape, monitored for proper quality and loudness level, and manual or automatic adjustment of the content can be performed as it is captured by the server or another tape machine.

Powered monitor speakers can be driven directly from the analog outputs of the AERO.qc, or an operator can simply use a pair of headphones.

AERO.qc In Baseband Ingest Application (SDI)

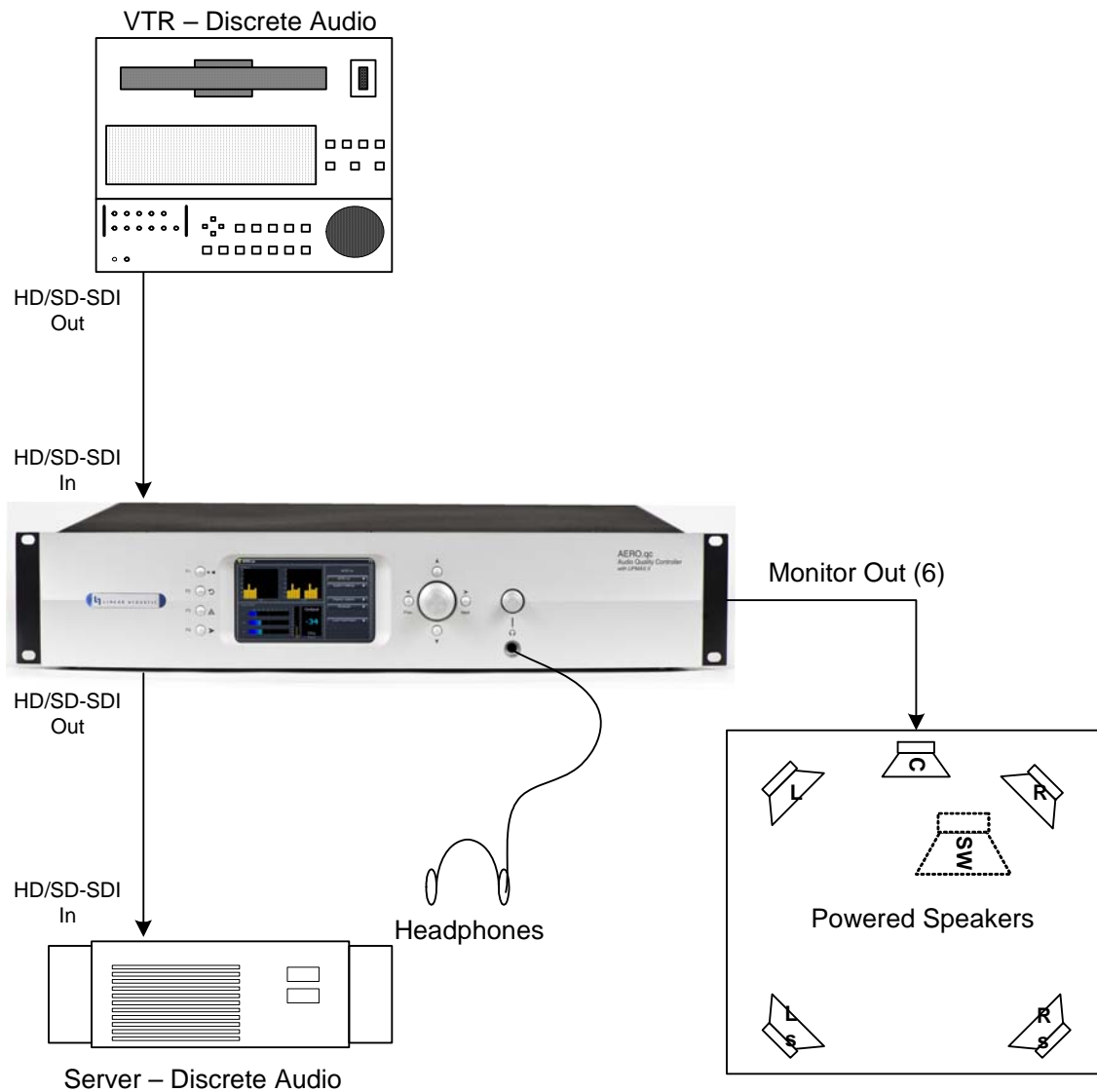


Figure 3-2 AERO.qc Used for ingest of SDI embedded audio channels

Similar to Figure 3-1, SDI audio can be played back from tape, monitored for proper quality and loudness level, and manual or automatic adjustment of the content can be performed as it is captured by the server or another tape machine. The AERO.qc also simultaneously outputs the audio via AES for other applications.

Powered monitor speakers can be driven directly from the analog outputs of the AERO.qc, or an operator can simply use a pair of headphones.

AERO.qc In Dolby E to Baseband Ingest Application

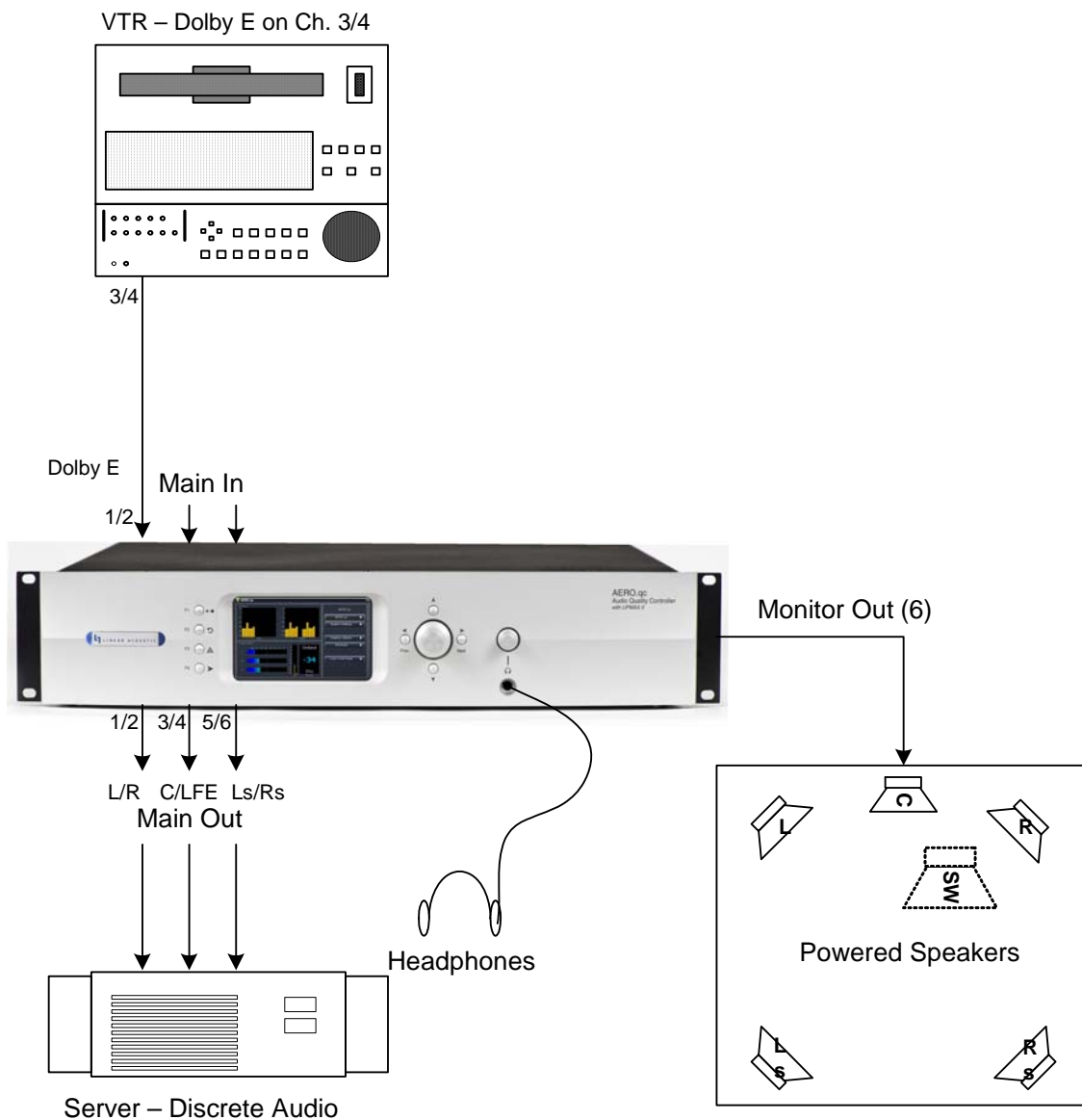


Figure 3-3 AERO.qc used for decode of Dolby E audio and loudness measurement during ingest

The AERO.qc can optionally contain a Dolby E/Dolby Digital decoder enabling it to replace several pieces of equipment normally used during ingest. Audio can be decoded, checked for proper loudness, adjusted if necessary, upmixed if desired, then output at discrete or embedded.

The auto-sensing capabilities of the unit allow it to easily handle either Dolby E or Dolby Digital with no re-patching or equipment changes.

AERO.qc In Live Application

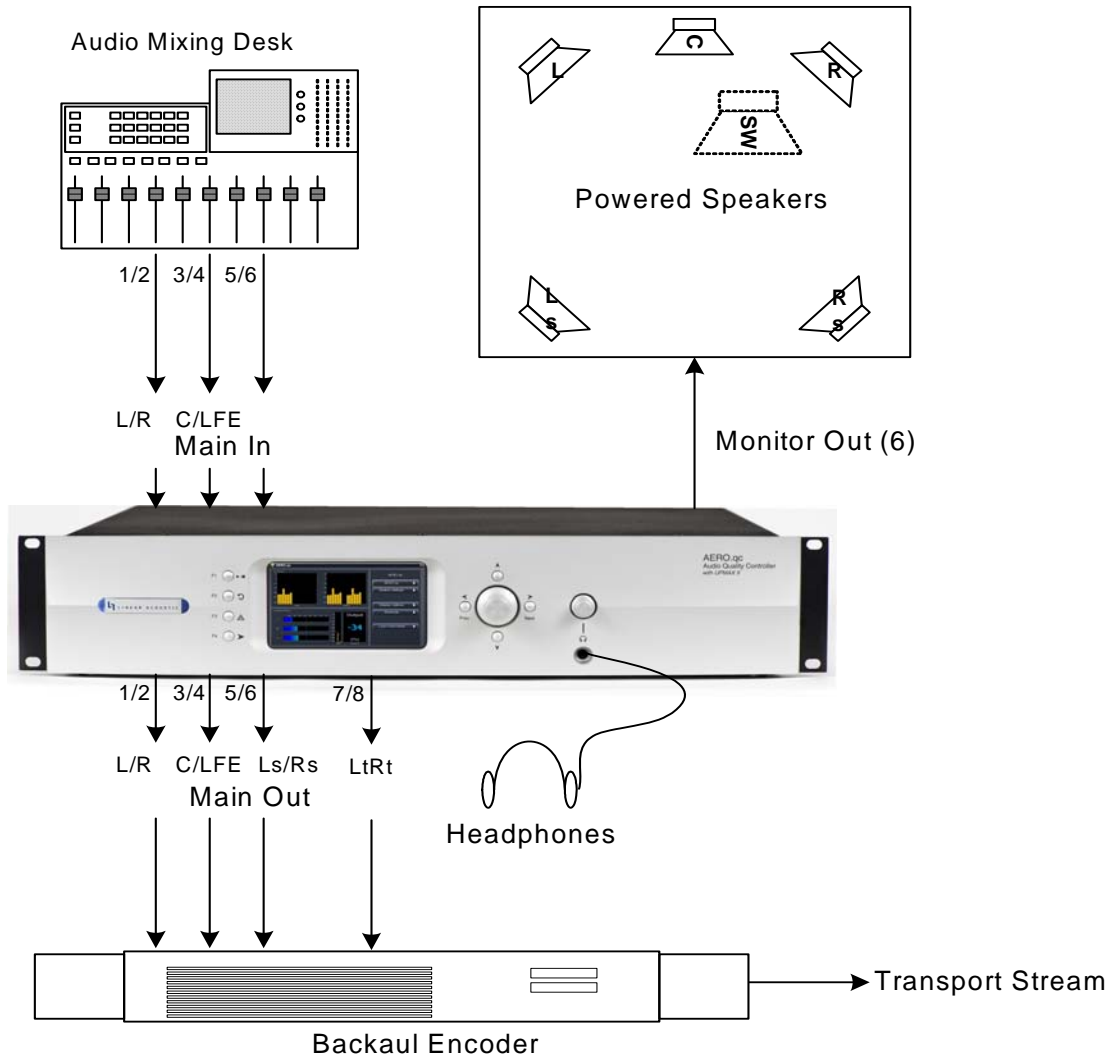


Figure 3-4 AERO.qc used for live production

The AERO.qc can be used during live production to provide metering, bus compression or peak limiting, and monitoring functions to enable verification of downmix. The unit can be used separately as an upmixer/loudness controller and monitor section since these two parts are separate functions.

Figure 3-4 is a greatly simplified live application. The AERO.qc is used to measure the loudness of the live mix and feeds resulting audio to a backhaul encoder/multiplexer which supports discrete audio via internal audio encoding (i.e. Adtec or Tandberg/Ericsson). This setup allows a quick overall audio measurements and adjustments to be made if necessary and dual power supplies and relay bypass make it ideal for live use.

Chapter 4: Detailed Operation

This chapter discusses in detail the structure of the Linear Acoustic AERO.qc, how to use the front panel interface to access the menus, accessing and saving factory and user processing presets, GPI and Metadata setup, and information about test tone generation features of the unit. Definitions of the processing presets and individual adjustments are covered in detail in Chapter 7.

4.1 Menu Navigation

Most of it is rather obvious, and a bit of experimentation will quickly make you comfortable navigating through the submenus. If in doubt, use the Left Arrow to back out towards the Main menu. The menus and submenus are structured to access more complex functionality as you go deeper into the hierarchy. See page 31 for a menu tree.

The left side of Figure 4-1 shows the main screen in *Locked* mode. Once unlocked, the main screen looks like the right side of Figure 4-1.

NOTE: The screen can be customized to provide exactly the desired display. The left side of the screen can show one or several metering screens and “Meters Only” mode allows for full-screen meter viewing.



Figure 4-1 Locked Main Screen (left); Unlocked Main Screen (right)

4.1.1 AERO.qc

This is the top-level menu for all audio adjustments. The sub-menus and functions are described below.

Main Settings

- **Main Settings** - Quick access to the following:
 - **Speaker Volume** (software)
 - **Headphone Volume** (Software)
 - **Bypass Dynamics**
 - **Upmix Enable**
- **Input Selection** - Quick access to the following:
 - **Main 5.1 Input** - Enables channels 1-6 as input; Default = On
 - **L2/R2 Input** - Enables channel 7/8 to be used as input; Default = Off
- **Output Selection** - Adjust channel pair utility delays and if Downmix should be sent to channel 7/8 output.
- **Display Options** - Adjust the type and quantity of left-side displays. For example, the unit ships with I/O Meters and Loudness Meter selected. These can be changed to taste. Full-screen metering can also be selected here. Below is a list of available meter types:
 - **DRC Meters** - displays gain reduction meters from the DRC section
 - **I/O Meters** - displays the 16 input channels (Main 1-8, Aux 1-8) and the 16 output channels (Main 1-8 and Monitor 1-8).
 - **Monitor Out** - displays the input and detailed output of the monitor section
 - **Oscilloscope**
 - **RTA**
 - **FFT Spectrum**
 - **Loudness Meter** - Quad ITU-R BS.1770 meters to show **F**ast, **M**edium, and **S**low results simultaneously. The fourth meter is the large numeric readout of LKFS loudness. Default integration times for meters 1-4 are 3 seconds, 10, seconds, 30 seconds, and 10 seconds respectively.

NOTE: *Detailed Loudness Meter Settings are purposefully tucked into the Dynamic Range menu to prevent accidental misadjustment. From the top menu, go to AERO.qc, then Dynamic Range, then all the way to the bottom for Loudness Meter where standard settings will be found.*

- **Off** - Disables the meter for that display.
- **Shortcuts** - Quick path to the following:

UPMAX II

Upmixing is the process whereby two channel audio can be re-mapped into 5.1 channels using level and phase detection techniques. The proprietary, industry standard and road-proven UPMAX™ algorithm is incorporated in the AERO.qc. This allows for realistic 5.1 channel audio to be created from normal stereo inputs, and performs even better when presented with matrix surround encoded two channel audio (i.e. LtRt). Two channel mono can also be accommodated in an audibly pleasing manner.

Critically, the UPMAX algorithm is fully downmix compatible. Many viewers are presented with Dolby Digital (AC-3) encoded audio that is downmixed in their set top box or receiver and they happily listen in stereo, none the wiser that more information is actually present. Upmixing systems that use phase shifting techniques can produce unacceptable results in these systems obscuring dialog and amplifying background artifacts.

UPMAX-II produces a very high quality, stable 5.1 channel result from a two channel stereo or surround encoded input signal. Built on the solid performance of original UPMAX, the new UPMAX-II uses both time and frequency domain techniques to produce more spacious results under many conditions. The results are downmix compatible producing excellent results in stereo and perfect reconstruction with standard broadcast-style downmixes delivered to most consumers.

Under many conditions, UPMAX and UPMAX-II sound similar. Stereo LoRo content will produce open and enjoyable results, and stereo LtRt content will result in a wide, engaging surroundfield. If content is produced while monitoring through UPMAX-II, stunning results can be obtained.

Uniquely, UPMAX-II does not rely on detection of anti-phase audio to produce the surround channel signals. This is of critical importance today because of the widespread use of low bitrate audio coding for delivery of some programming. This can sometimes cause coding artifacts to be unmasked and reproduced by the surround channels. UPMAX-II avoids this by its very design and unmasked artifacts are minimized or avoided completely.

Auto mode enables control of the upmixing function of the AERO.qc in situations where there is no GPI or metadata control signal available. This is accomplished by detecting if the RMS signal level in the Center or Surround channels crosses an adjustable threshold.

If no signal from these channels crosses the threshold, the upmixer is set to “Auto” mode where the Left Front and Right Front channels are upmixed and the resulting upmixed signal is combined with the discrete Center, LFE, and Surround channels.

If signals in any or all of the discrete Center or Surround input channels cross the threshold, the Auto upmixing process is inaudibly crossfaded to bypass, and all audio channels are then passed completely discrete with no upmixing at all.

This method absolutely prevents clipping or cutting dialogue while ensuring that two channel content remains upmixed to provide viewer expected surroundfield consistency.

- **Load / Save Preset** - Recall or store a preset for the Upmixing section.
- **Main Settings**
 - **Power** - Inserts or removes the entire upmixing engine from the signal path. This is not a crossfaded switch and enabling or disabling could make an audible tick. To quickly enable or disable upmixing, use the Upmix Enable control (next)
 - **Upmix Enable** - Turns the upmix process on or off without removing it from the signal path.
 - **Auto Mode** - Allows automatic control of upmix enable/disable based on the detection of audio in the Center, or Left Surround, or Right Surround Main Inputs. Detection is valid if audio is present above threshold in any of these three channels.

NOTE: *When Upmix is Enabled and Auto Mode is active, the Center, LFE, Ls, and Rs channels (3-6) are passed through at all times - there is no muting and thus no chance for loss of dialogue. If Auto Mode is NOT active, Center, LFE, Ls, and Rs channels (3-6) are MUTED!*

- **Mode: UPMAX II/UPMAX II LL/UPMAX I** - Selects between industry standard Linear Acoustic UPMAX I and UPMAX-II, both of which produce very consistent and downmix compatible results. UPMAX II has longer latency and so there is a lower latency version provided for experimentation called UPMAX II LL.
- **Center Extraction** - A checkmark indicates that the Center channel output of the upmixer is being reproduced from the Center channel Main output. Unchecked indicates that Center channel audio from the upmixer is re-combined back into the other channels (no audio is lost). Default is **Enabled**
- **Surround** - A checkmark indicates that the Surround outputs of the upmixer are being reproduced from the surround Main outputs. Unchecked indicates that Surround outputs of the upmixer are re-combined back into the other channels (no

audio is lost). Default is **Enabled**.

- **LFE Generation** - A checkmark indicates that an LFE channel is being created to produce the 0.1 channel of the upmixer. Default is **Enabled**.
- **LFE When Off** - A checkmark indicates that an LFE channel will be created for input audio in auto mode when upmix has transitioned to off. Default is **Disabled**.
- **Adjust**
 - **Center Exclusivity** - Active in UPMAX-II Only. Adjusts the degree to which center channel information is excluded from the remaining channels, especially the surround channels. Use with care at extreme ranges.
 - **Upmix Center Width** - Controls how much of center channel output of the upmixer is spread back into the Left and Right channel Main Outputs of the unit. 100% width equals all Center channel information summed back into the Left and Right outputs, while 0% equals all Center channel information sent to the Center Channel output; Default is **33%**.
 - **5.1 Center Width** - Controls how much of center channel output of the discrete input is spread back into the Left and Right channel Main Outputs of the unit; Default is **0%**.
 - **Center Level (UPMAX II Only)** - Allows minor adjustment of upmixed Center channel level.
 - **LFE Level** - Controls how much of a 80Hz low pass filtered sum of the Left, Center, and Right outputs of the upmixer is fed to the LFE channel Main Output; Default is **-18dB but please adjust to taste with proper monitoring**.
 - **Dimension** - Controls how far Front Left and Front Right are pulled back into Left and Right Surround channels. the surround channels are panned to the back speakers. 100% sends all LfRf upmix audio to Ls and Rs. Use with extreme caution and monitor the resulting downmix! Default is 0%.
 - **Surround Depth** - Controls how far the surround channels are panned to the back speakers. 100% sends all surround channel upmix audio to Ls and Rs, while 0% returns that audio to the front channels. Default is 100%.
 - **Surround Width** - Controls the amount of separation between the upmixed Ls and Rs signals where 0% equals mono and 100% equals stereo. Note that exceeding 100% is possible and should be used with care while checking for stereo downmix compatibility.

- **Auto Mode Settings**
 - **Detection Thresh** - Sets the RMS point at which Auto upmixing is disabled allowing discrete audio to pass untouched. Adjustable from -90 dB FS (very sensitive, switches to discrete with little more than noise present) to Off (not sensitive at all, will never switch to discrete). Default is **-75 dB**, but adjust to match facility signal levels (should produce a definite switch at transitions)
 - **Discr -> Up** - Sets the timing for engaging the upmixer. It is important that this happen quickly enough to prevent two channel material from not being upmixed or the transition from discrete to upmix being audible. Normal default is **Quick**, but adjust to taste.
 - **Up -> Discr** - Sets the timing of the transition from Upmix back to discrete. As the original audio is not ever muted, this transition can be rather slow without being noticeable thus preventing audible false switches. Default= **Slow**
 - **LFE When Discrete** - A checkmark indicates that an LFE channel will be created for input audio when the upmixing section has been turned off. Default is **Disabled**
 - **Detect Center** - Selects whether the Center channel main input (channel 3) is used for determining whether to upmix while in Auto mode. Normal mode is Enabled, but for L/R + Announce mixes, detection should be disabled.

NOTE: *Detect Center should be DISABLED for 3-channel L/R + Announce mixes. This will allow the announce audio on channel 3 to mix with the upmixed version of channels 1/2 without causing the upmixer to automatically stop.*

Dynamic Range

Please see chapter 7 for details.

Speaker Output

Extensive control of level, equalization and delays for each of the analog monitor outputs. Presets for monitor setting can be saved and recalled here as well. This is also where the Patch Point (source) for the monitor section is selected as well as any downmix or matrix decoding for emulation.

NOTE: *Think of Patch Points in terms of a traditional patch bay with points located on I/O and within the signal path. Monitor and Headphone outputs can be moved almost anywhere in the chain.*

- **I/O** - Monitors the main I/O of the unit:
 - **Main 5.1 Input**
 - **Main 5.1 Input Downmixed**
 - **Main L2/R2 Input**
 - **Calibration Mic (not used)**
 - **Main 5.1 Output**
 - **Main L2/R2 Output**
- **Interim** - Monitors post upmixer audio
- **External Inputs** - Monitors External (Monitor) Inputs:
 - **External 5.1 Input** - Monitors External 1-6 as a 5.1 channel signal
 - **External 1/2 Input** - Monitors External 1/2 as a stereo source
 - **External 3/4 Input** - Monitors External 3/4 as a stereo source
 - **External 5/6 Input** - Monitors External 5/6 as a stereo source
 - **External 7/8 Input** - Monitors External 7/8 as a stereo source
- **Sidechains** - For listening to selected control sidechains within the DRC section
- **Multiband Solo** - For listening to selected bands within the multiband DRC section.
- **I/O** - Monitors the main I/O of the unit:
 - **Main 5.1 Input**
 - **Main 5.1 Input Downmixed**
 - **Main L2/R2 Input**
 - **Calibration Mic (not used)**
 - **Main 5.1 Output**
 - **Main L2/R2 Output**

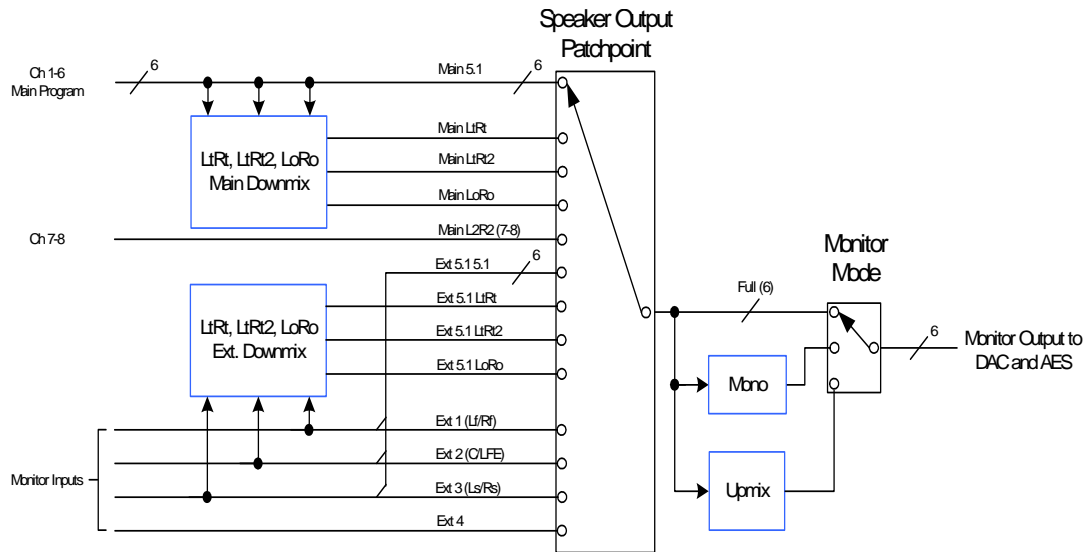


Figure 4-2 Monitor Source Select

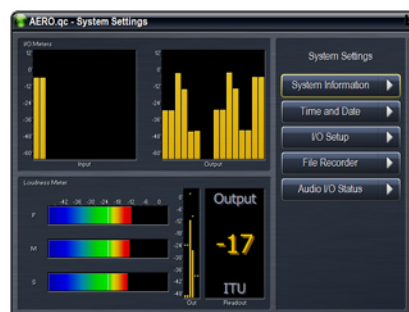
Please see chapter 5 for more details on room calibration.

Headphone Output

Extensive control of headphone output including Patch Point, loudness compensation (works only when software volume control is used), and equalization. Presets can be stored for different users and different headphone types.

4.1.2 System Settings

The **System Options** menu provides the following functions:



- **System Information** - Find system information such as software and firmware versions, set password, and adjust HTTP access in this menu.
- **Time and Date** - Allows configuration of the AERO.qc internal clock for correct

time zone, date, and time. It also allows enabling a Network Time Piece (NTP) service which can automatically connect to a time server and keep time and date synchronized. Depending on the network to which the AERO.qc is connected, configuration of the facilities Ethernet router may be necessary

NOTE: The AERO.qc has DHCP enabled which means that it will automatically fetch an IP address from the network. To assign a static IP address, the units MAC address is provided and the local router can be configured to map the desired IP address to that unique MAC address.

- **I/O Setup** - Adjust inputs, outputs and clock source here:



- **H/W I/O Setup** - Select the source for each of the ten processing core inputs:
 - **In 1/2 = AES In 1/2** (SDI 1/2 - 15/16)
 - **In 3/4 = AES In 3/4** (SDI 1/2 - 15/16)
 - **In 5/6 = AES In 5/6** (SDI 1/2 - 15/16)
 - **In 7/8 = AES In 7/8** (SDI 1/2 - 15/16)
 - **In 9/10 = AES In 9/10** (SDI 1/2 - 15/16)
 - **Clock Ref** - Select between Internal 48kHz, AES Reference In, AES In 1, and SDI.

-
- **SDI Embedding** - Selects the source for each of the 16 possible audio channels for embedding (first and last shown, defaults in bold):
 - **Out 1/2 = AES 1/2** (AES 3-10, Enc A, Enc B, SDI 1/2 - 15/16, Mute)
 - ...
 - **Out 15/16 = SDI 15/16** (AES 3-10, Enc A, Enc B, SDI 1/2 - 15/16, Mute)

NOTE: *If SDI Embedding is enabled, clock source will automatically switch to SDI.*

- **SIMM Source (Option)** – Selects input for Dolby Digital/Dolby E decoder SIMM module. Allows selection of any AES or SDI input.
- **GPI** – TBD
- **File Recorder** – Allows Input, Output, or Input+Output to be recorded to a USB drive for troubleshooting purposes. Audio can be recorded to an installed USB thumb drive. Note that this is currently intended for factory diagnosis of audio issues and requires a special player for proper reproduction. With instructions from the factory, 6 or 8 channels of input and/or output audio can be captured and emailed to us to assist with troubleshooting.

Chapter 5: Troubleshooting

The Linear Acoustic AERO.qc is a very stable and reliable unit, and most problems can be traced back to mis-wiring causing incorrect signals to be applied to the unit, or more than likely mis-configuration. As simple as the unit is to use once installed, it is very flexible and thus a necessarily complex processor. In an effort to speed troubleshooting, some common problems and solutions are described below.

5.1 Problems and Possible Causes

One of the best troubleshooting features of the AERO.qc is the hard-bypass of the audio signals. This is useful because it allows instant removal of the unit from the signal path. Hard-bypass can be accomplished two ways, the simplest being to remove AC power from the unit by turning the power switch off. A less dramatic way to accomplish the same result is to access the System Options menu and enable Master Bypass. If you are unsure of what is happening in a particular system, simplifying the signal path is a good start and will help isolate problems quickly.

5.1.1 Unit won't power on

Check to make sure that the unit is connected to a live outlet (it happens...). If proper AC voltage is being applied to the unit, remove power and check the fuse (the IEC inlet module has a fuse compartment that can be carefully opened with a small screwdriver). If the fuse is blown there is a spare in the holder. Install the replacement fuse, re-insert the fuse carrier, and re-apply power.

NOTE: *Replace fuses only with the same size and rating: 5mmx20mm T1AL (250V)*

With the unit plugged in to a valid AC power source, press the Left, Up, and Right positions on the RIGHT navigation ring (i.e. a thumb on the 11 o'clock and 2 o'clock positions). The Linear Acoustic logo blue LEDs will illuminate first and the unit should then power on normally.

NOTE: *This may take some pressure as it is activating three buttons, but with a little persistence and patience you will quickly get used to it.*

5.1.2 The "LA" symbol is flashing red

The small Linear Acoustic logo in the upper right corner of the display is used to indicate that the reference input is either missing or wildly incorrect. Check to make sure that an AES reference signal is connected to the unit. Note that this signal can be either AES black

(i.e. an AES signal with silent audio), or an AES signal with audio. The unit will NOT accept word clock or video reference signals. Note that in addition to the AES Ref In, reference can be derived from Main Input 1, SDI and Internal within the I/O Options menu (see Chapter 4).

5.1.3 Output Audio Clicks and Pops

This could indicate that the AES Reference is missing or at the wrong sample rate (the unit expects to be locked to 48kHz). The unit is designed to default to an internal 48kHz reference in case the external reference is removed. This allows audio to continue, but due to the sample rate converters present on each input pair, the outputs will be asynchronous with the inputs. The solution is to make sure that the reference is connected.

It could also mean that audio applied to the unit is not properly referenced, or that the equipment that the AERO.qc is feeding is not properly referenced. Using the master bypass function of the unit is a helpful way to troubleshoot this issue.

5.1.4 Audio Sounds Strange

Reports of audio that sounds like it is in a tunnel indicate timing issues between the channel pairs. Most often viewers watching in stereo will be the first to notice such problems. Some suggestions include making sure that the delay settings in the AERO.qc are all equal (if they are intended to be so). Offsets as small as a couple of milliseconds may quickly become audible under the right conditions.

Another suggestion applies to those who decode network audio using a Dolby DP564 AC-3 decoder. There is a monitoring mode that allows the analog and digital outputs to be used. This can inject 10msec of delay into the surround channel audio which will cause complaints. The correct setting is either to have the unit in so-called “Bypass Mode” or to set the monitoring mode to “Digital.”

5.1.5 Audio Issues Disappear In Bypass

This can be caused by a number of things, including the fact that a processor like AERO.qc can apply a great deal of gain to incoming signals. Things that would not normally be audible quickly get pulled out of the background.

We recommend taking a good hard listen to station audio. Hums, buzzes, clicks, pops, phase issues, frequency response problems, bass build-up on the news set, etc... will all be amplified in varying degrees and can ruin otherwise excellent programming.

As with all traditional processing, the input audio should be immaculately clean else it might be made worse- and the difference now is that it will sound bad in surround!

5.1.6 Audio Fixed by Re-boot

This can be a very misleading situation. Experience has shown that re-booting can sometimes fix things by interrupting the AES signals thereby causing equipment downstream to re-lock to the incoming signals. This will sometimes mask the true problem, and a better method would be to select a minimal processing preset (such as Protection Limit) and observe the results. Careful and methodical troubleshooting is essential.

5.1.7 Audio Has Artifacts

Sometimes described as “squirrels in the surrounds” this is almost always caused by source content that has been encoded at too low a rate. For example, some program delivery services or server systems may use MPEG 1, Layer 2 audio data reduction to save bandwidth. When operated at 256kbps or higher per stereo pair, there will be no issues. However, 128kbps or lower can result in coding artifacts that are present in the original stereo and may be unmasked by upmixing. Note that this content is inferior in quality and will have similar results without upmixing if consumers matrix decode the signal (which is the default mode for most home theatre A/V receivers). Our best advice is to take this up with the program provider or server manufacturer. Today, bandwidth is cheap and there is no good reason to run lower than 256kbps at a minimum.

- NOTES -

Chapter 6: Monitor Setup

6.1 Monitor Calibration (Analog Outputs)

The Linear Acoustic AERO.qc is supplied with a monitoring grade digital-to-analog converter that provides balanced outputs of all eight audio channels. The level of these outputs can be controlled by the front panel rotary encoder or from an external volume control (a higher quality “big knob” remote is available from Linear Acoustic).

The individual channels can be calibrated over a 10-dB range to allow matching the unit to power amplifiers or powered speakers. To maintain the lowest noise and widest dynamic range, it is important to manage analog gain stages properly and in order.

It is strongly recommended that gains be adjusted in the following order:

- Gains of power amplifiers or powered speakers
- AERO.qc channel trims
- AERO.qc volume control

With an input level of -20dBFS, the analog outputs will be at +4dBu. While appropriate for most power amplifier/speaker combination, many powered loudspeakers will produce excessive output SPL when driven at this level and thus will usually contain continuously adjustable input gain controls for trimming. In some cases, these might not have enough range or the gain controls may just be coarse adjustments. In this case, it will be necessary to use an external attenuator to match the reference levels of the AERO.qc and the powered speaker or amplifier. Contact the factory for more information.

6.1.1 Main Channel SPL Adjustment

Monitor calibration is the single most important part of calibrating an AERO.qc and a proper mix environment. Take the time to get this right and check it regularly.

With the AERO.qc properly referenced and volume and channel trims at 0dB, apply pink noise at plant reference (such as -20dBFS) to the Center channel (Channel 3). This is best accomplished via the Test Tones menu (under System Options).

NOTE: *It is very important to make sure that Volume is set to Reference. This can be done via the front panel under the Shortcuts menu, or on the available GPI remote control where Reference position is indicated by a small green LED below the monitor knob.*

Using a sound pressure level (SPL) meter set to “C-weighted” and “Slow Response” adjust the Center channel power amplifier to the desired SPL. Typical industry levels range from 78-79dB/C/Slow for near field television mixing to 85dB/C/Slow for far-field film mixing. It is important to note that while an overall reference level is very important, it is more important to ensure that all main channels are set to the *same SPL*.

If the power amplifier or powered speaker is adjusted as low as possible and the desired target SPL cannot be reached, use the Center channel output trim on the AERO.qc to adjust to the desired SPL. This adjustment can be found under the Speaker Output -> Calibration -> Levels menu.

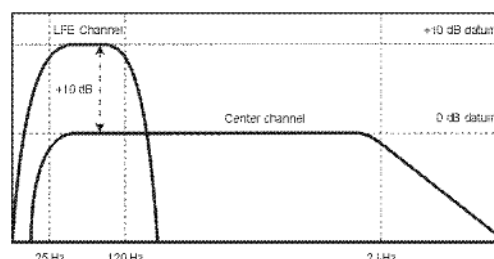
Repeat this procedure for all of the main audio channels, again making sure that the AERO.qc volume control remains at reference (green LED on). This allows the “return to reference” function to support any desired volume setting while mixing, and a simple return to the calibrated SPL during final mixing or mastering.

6.1.2 Subwoofer SPL Adjustment

For 5.1 channel mixing, the subwoofer is required to apply 10dB of gain to the signal applied to it. This means that if the reference SPL of the main channels is set to 79dB SPL, then the subwoofer is set to 89dB SPL. This is based on the long-standing practice in film to allow an extra 10dB of headroom in the Low Frequency Effects (LFE) channel and was originally required by optical and magnetic tracks that could not pass high-level low frequency information without severe overload. Modern 5.1 systems no longer have a headroom issue, but for compatibility with industry practice and legacy material, the increased reproduction level of the subwoofer is maintained.

It is strongly recommended that this adjustment be made with the real-time analyzer (RTA) built into the AERO.qc and not an SPL meter due to constraints in low frequency response performance of typical meters. The procedure is simple: after calibrating the level of the main channels, apply pink noise to the Center channel and note the average level. Next, apply pink noise through the LFE channel to the subwoofer and adjust its level so that it has about 10dB of additional mid-band level as compared to the Center channel.

The desired relationship between Center and Subwoofer is shown below



Note that this is a real-time analyzer display of the response of a typical film mixing stage. Due to the large distance from the speakers to the mixer, there is a natural roll off above 2 kHz (described in SMPTE 202M, and sometimes called the "X-curve.") In small mixing and control rooms this response would be flat.

6.1.3 Speaker Equalization

The AERO.qc provides basic tools for calibrating the frequency response of monitor speakers: noise generation coupled with a real-time spectrum analyzer that has an external analog input. Each of the speaker output channels has 12 bands of adjustable parametric equalization, each with level, "Q" or bandwidth, and frequency. Each band is full-frequency response allowing very flexible calibration. Delay can be added on each individual channel to compensate for arrival time differences.

To calibrate a room, enable pink noise on a given channel. Pink noise can be generated internally (System Options -> I/O Options -> Test Tones) and output this noise channel-by-channel basis. Connect a microphone to a preamplifier and then to the Cal. Mic Input connector of the AERO.qc. Enable the RTA under Speaker Output -> Calibration -> RTA and set it for 1/3 octave, with averaging around 30. Make sure the RTA input is set for MIC.

Enter the parametric EQ for the channel by going to Speaker Output -> Calibration -> Parametric EQ. The trick is to use as few bands as possible and work on problem areas first. The goal is a flat frequency response. Do not be tempted to over-eq, these are average measurements and it will never be perfect. Also, it is not possible to equalize out room problems. If you run out of bands, you may need to resort to explosives.

6.1.4 Speaker Delay

In a typical room setup, some speakers will be further away than others. If we assume that it takes about 1msec for audio to travel 1 foot (1/3 meter), then a quick drawing of the speaker setup will reveal which speakers need to be delayed. Please note that the delays included in AERO.qc are both positive and negative. Positive delay actually applies negative delay to all the other channels (we have not reinvented time. yet.).

For example, if you measure from the desired mix position to the center speaker and it is 10 feet, while Lf and Rf are 12 feet and Ls and Rs are 15 feet, the rough adjustments would require adding 5msec to Center, and 3msec to Lf and Rf.

6.1.5 Listen!

Once the alignment of all channels is complete, play some content through the system and *listen* to make sure things sound normal. Never underestimate the effectiveness of taking a listen using known content to make sure that things are really what they seem. A favorite SACD or DVD is a good start, we suggest the sublime James Guthrie remasters of Pink Floyd's Dark Side of the Moon, Wish You Were Here, or the Wall.

Chapter 7: Presets & Processing Adjustment

The Linear Acoustic AERO.qc contains multiple factory-programmed processing presets that have been developed after many hours of listening and experimenting, and hundreds of program sources across all genres were used during this tuning. Preset creation is an on-going process, and we regularly implement presets designed by customers for specific applications.

Of course, all processing controls are accessible and users can custom design special presets that might be even more appropriate for a given situation. We strongly recommend starting with the factory preset that is closest to the desired objective, then fine tuning it to reach the desired goal. This will minimize the troubles that will likely be encountered as many adjustments interact.

7.1 Description of Factory Presets & Adjustments

Each unit ships with a number of presets that are useful for most situations. A brief description of each is given below. Following the description of presets, a description of each control available for use by all presets is given. Note that not all presets use every feature. For example, only presets labeled as “NR” will use the multiband source noise reduction feature.

Factory Presets

- **TV 5B GEN - TeleVision 5-Band GENeral.** This is the most commonly used preset. It provides a moderate degree of dynamic range processing, and is appropriate for all types of content. It is the factory default for all core preset choices. Use of this preset is highly recommended as it will produce audio that will have an average dialog loudness of -27 as measured by a Linear Acoustic LQ-1000 or LAMBDA.
- **TV 5B LIGHT - TeleVision 5-Band LIGHT.** This is very similar to TV 5B GEN, however the ratio of the multiband compression has been reduced closer to 2:1 for a more gentle action. Note that this will result in audio that strays further from an average dialog loudness setting.
- **TV 5B HVY - TeleVision 5-Band HEAVY.** Similar to TV 5B GEN, however the ratio of the multiband compression has been increased for a more dense and less dynamic sound.
- **TV 5B LIGHT - Television 5-Band LIGHT:** Similar to TV 5B GEN, however the ratio of the multiband compressors has been decreased for a more open sound. Please be aware that the lessening processing lessens the ability of the unit to control loudness tightly to a given target.

- **TV 5B LOUD** - **TeleVision 5-Band LOUD**. Similar to TV 5B HVY, but louder and more punchy.
- **ITU LOUDness LMT** - Utilizes a specially tuned Input AGC plus Multiband Limiters and the Final Limiter to slowly adjust the average program loudness to a given value and the multiband and final limiters will act until the AGC catches up. This preset is appropriate for ingest or live applications but because the Multiband AGC is bypassed it has less ability to manage spectral balance which is important for transmission.
- **Protection Limit** - Sets all processing to unity ratio (1:1) and leaves only protection look-ahead peak limiters in the signal path. Master output level is also set to 0dB so the processor will provide a unity gain path for applied signals.

7.1.1 Processing Structure

Before choosing to jump into shark infested waters, we *STRONGLY* recommend starting with a factory preset and modifying from there. Preset creation starting from scratch is an incredibly time consuming process requiring large amounts of time and huge amounts of content spanning all genres including music of every type, films of every type, television dramas and sitcoms of every type, and talking-head programs of every type. We have already done much of the heavy lifting, and provide presets based on our own listening and feedback from customers over the past few years. Certain parameters such as crossover frequencies and channel coupling are not adjustable and are hard-coded to the most appropriate values. That being said, access to all adjustable parameters is provided. There are no factory adjustments hidden from users. Be careful with this power as most all settings interact with each other, sometimes in ways that might not be immediately audible.

Below is a drawing showing the general signal flow of the processing core to indicate what part of the chain is being adjusted by each parameter. Note that this signal flow is also captured in the “Adjust Processing” menu, with the top of the list being the input, and the bottom of the list being the output.

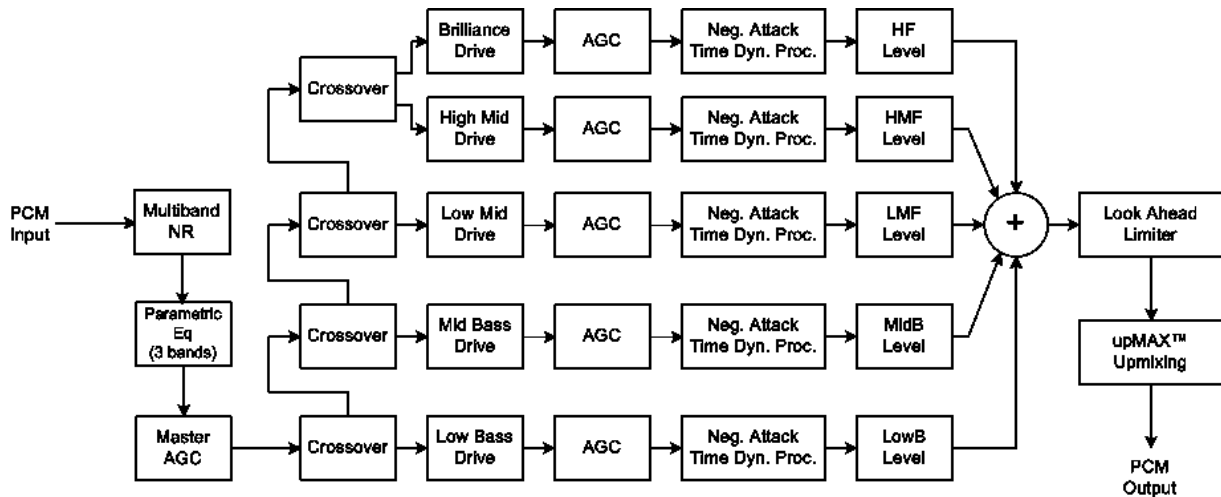


Figure 7-1 General signal flow of the AERO.qc processing core

7.1.2 Adjusting Processing

Following is a description of all adjustable core processing parameters.

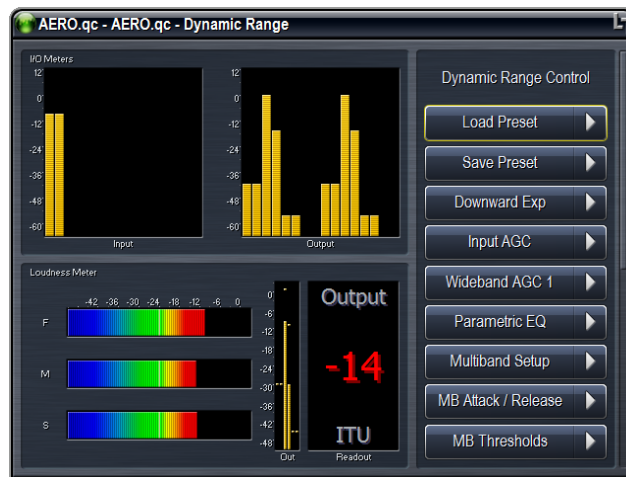


Figure 7-2 Dynamic Range Control Menu

But first... A Word on Our Crossovers

The multiband crossovers in the AERO.qc consist of second order Linkwitz Reilly style filters that hard coded to specific frequencies. As the processing required for television applications is not as aggressive as other mediums, little advantage can be gained from changing these values, and the remainder of the processing relies on these characteristics remaining constant.

For reference, the crossover frequencies are:

- Band 1 (Low Bass): 20 Hz - 60 Hz
- Band 2 (Mid Bass): 30 Hz - 200 Hz
- Band 3 (Low Mid): 170 Hz - 1.15 kHz
- Band 4 (High Mid): 950 Hz - 6.1 kHz
- Band 5 (Brilliance): 5.2 kHz - 24 kHz

While the multiband AGC section allows the number of bands to be varied between four and seven, five bands has been found to be a good match for television audio. More bands can be selected if dense processing is desired, and four bands can be used when source content reliably requires minimal processing.

Adjust Noise Reduction

This menu allows access to the multiband source noise reduction controls. Basically a multiband downward expander, source noise reduction can be helpful in situations where aggressive processing is desired, but the source audio contains substantial background noise. Most modern content does not require noise reduction and in general it is recommended that it only be enabled if necessary. Adjustable parameters are:

- Dwn Exp Rnge: 0dB to 24dB where 0dB equals no noise reduction. The default is 0dB.
- B1-B5 Threshold (Band 1 - Band 5) - Adjusts the point at which 2:1 expansion takes place, and is variable from -90dBFS to 0dBFS. The lower the setting (i.e. the closer to -90), the earlier the expanders will “open” and stop gating noise. Nominal settings are:
 - B1-B4: -40dBFS
 - B5: -50dBFS

Parametric Eq(ualization)

Three bands of parametric equalization are provided for fine tuning if necessary. None of the factory-supplied presets use the parametric equalizers, but they are provided to create notch filters or other effects if necessary. Each filter has a gain control with a +/-12dB range, a center frequency control that varies from 20 Hz to 22.050 kHz, and a bandwidth or “Q” control that varies between 0 and 10. Normal default settings for all bands are Gain = 0dB, in other words bypassed.

Input AGC

The input AGC is a very slow acting front-end gain control with a 36dB gain range whose only purpose is to make sure that the following processing stages are fed with the correct average audio levels. It is basically the automatic equivalent of an operator slowly riding a gain control on a console to keep the audio close to reference level. Wideband in nature, the AGC is not meant to perform rapid gain reduction or expansion as its actions will be more audible, as with all wideband gain processors. As a slow gain rider, its actions are nearly inaudible thanks to the multiband processing that follows it. The AGC has two stages of gating where the gain expansion is slowed or stopped to prevent background noise increasing.

Adjustable parameters are:

- Gating Thresh(old): 0dBFS to -90 dBFS

-Gating sets the point at which the AGC release time is made extremely slow to prevent increasing background noise and allow the AGC to return to unity gain.

- Freeze Thresh(old): 0dBFS to -90dBFS

-Freeze stops all gain change (i.e. when the audio drops to silence), and remains frozen at its current gain value until the threshold is exceeded.

NOTE: *Gate and Freeze **MUST** be adjusted to match plant practices. Very quiet audio (such as a golf match) benefits from having processing frozen when input audio drops below a given level to prevent “boosting the cricket” sounds.*

- Ratio: 1:1 to Inf:1

- Range: 0dB - 36 dB

-Range sets how much gain expansion above unity is performed, and this amount is subtracted from the total AGC gain range of 36dB, so the default value allows for 24dB of expansion and 12dB of compression. This adjustment is reflected in real time by changing the AGC meter scale.

- Threshold: -18dBFS - 0dBFS

- Attack: 0 - 150, slowest - fastest

- Release: 0 - 150, slowest - fastest

- Progressive Release: 0 - 100, slowest - fastest

-Sets the speed at which the release time is increased faster at very low gain values. This feature approximates a logarithmic release to help recovery from dramatic gain reduction more quickly.

Multiband AGC

This section is the heart of the dynamics processing engine. A multiband AGC (i.e. compressor) that allows for medium ratio (3:1 is default) adjustment of audio band:

- Number of Bands: Four/Five/Six/Seven Bands (default: five bands)

- Ratio: 1.0:1 - Inf:1

- B1-B5 Inf:1 Above Thresh: Enabled / Disabled

-AGC automatically increases ratio to Infinity:1 once a signal exceeds the threshold (set below), allowing for expansion below the threshold and limiting above the threshold. Useful for bass frequency control.

- Range: 0dB - 24dB

-Range sets how much gain expansion above unity is able to be performed. This adjustment is reflected in real time by changing the AGC meter scale.

- Progressive Release: 0 - 100, slowest - fastest

-Sets the speed at which the release time is increased faster at very low gain values. This feature approximates a logarithmic release to help recovery from dramatic gain reduction more quickly.

- B1 - B5 AGC Attack: 0 - 150, slowest - fastest

-How fast an input signal is acted upon once it crosses the set threshold

- B1 - B5 AGC Release: 0 - 150, slowest - fastest

-How fast an input signal recovers from a gain change once that signal falls below the set threshold.

- B1 - B5 AGC Thresh(old): -12.00dB - +12.00dB

-The reference point for the attack and release parameters to act on the audio signal present in that band.

- MB Gating Thresh(old): 0dBFS to -90 dBFS

-Gating sets the point at which the multiband AGC release time is made extremely slow to prevent unnecessarily increasing background noise and allow the AGC to return to unity gain.

- MB Freeze Thresh(old): 0dBFS to -90dBFS

-Freeze stops all multiband gain change (i.e. when the audio drops to silence), and remains frozen at its current gain value until the threshold is exceeded.

Multiband Limiters

Performs multiband limiting of the signals coming from the multiband compressor.

- B1 - B5 Lim(it) Thresh(old): +12.00dB - 0.00dB

-The point above which limiting action takes place at an Infinity:1 ratio.

- B1 Soft Clip Thresh(old): +12.00dB - 0.00dB

-The point above which band one (low bass) is soft clipped. This helps maintain a “tight” bass sound.

Multiband EQ

This is the section where each of the processing bands is summed and where overall frequency response can be tailored.

- B1 - B5 Out(put) Mix: -12dB - + 12dB

-Sets the mix level for each band summing all bands back together. These controls are prior to the final look ahead limiter and increasing gain may cause more final limiting (possibly more than desired)

Final Stage

This final section of the processor is where the final look ahead peak limiter and bass soft clipper are adjusted. The look ahead limiters are wideband, limited to 6dB of gain reduction, are extremely fast, and due to their look-ahead nature are virtually transparent even at full gain reduction. Their purpose is to control peaks that make it through the multiband section. Adjustable parameters are:

- Final Limiter Drive: -6dB - +6dB

-Sets the level at which the wideband sum of all bands is fed to the final limiter.

- Output Level: -36dB - 0dB

-Sets the output level for the current preset. Can be used to match the measured loudness of one preset to another. This is useful as more aggressive presets will measure differently from less aggressive versions.

NOTE: *This is the control used to adjust overall loudness of a facility. An ITU-R BS.1770 compliant loudness meter should be used with the goal of setting the loudness to match either the facility target or the dialnorm value in metadata.*

NOTE: *As an example, with TV 5B Gen selected, normal dialogue-based programming applied, and Output Level set to -11dB, loudness will measure approximately -27dBLKFS using a Linear Acoustic LAMBDA or Dolby LM100.*

Upmixing

Upmixing is the process whereby two channel audio can be re-mapped into 5.1 channels using level and phase detection techniques. The proprietary, industry standard and road-proven UPMAX™ algorithm is incorporated in the AERO.qc. This allows for realistic 5.1 channel audio to be created from normal stereo inputs, and performs even better when presented with matrix surround encoded two channel audio (i.e. LtRt). Two channel mono can also be accommodated in an audibly pleasing manner.

Critically, the UPMAX™ algorithm is fully downmix compatible. Many viewers are presented with Dolby Digital (AC-3) encoded audio that is downmixed in their set top box or receiver and they happily listen in stereo, none the wiser that more information is actually present. Upmixing systems that use phase shifting techniques can produce unacceptable results in these systems obscuring dialog and amplifying background artifacts.

There are very few controls required for setup of the upmixer:

- **Enable Center:** On sends center channel audio to the Center output; Off keeps it in Left and Right.
- **Enable Surround:** On sends surround information to the Surround Outputs; Off keeps it in Left and Right.
- **Enable LFE:** On sends bass enhancement audio to the LFE channel; Off disables bass enhancement.

- **Center Width:** controls how much of center channel output of the upmixer is spread back into the Left and Right channel Main Outputs of the unit; The Default is 33% to prevent two channel mono material from being reproduced as stark center channel only audio, and to prevent center channel build-up with music content.
- **LFE Level:** Sets the level of the created bass enhancement signal sent to the LFE output.

AutoMAX-II™ Automatic Upmixing

Upmixing is performed on the Lf/Rf signals, and the output of the upmixer is summed with signals present on the C/LFE and Ls/Rs inputs. This is useful primarily for situations where no GPI contact closure and no metadata commands are available and two-channel and 5.1 channel audio are present within the same three AES pairs and it is impossible to explicitly signal when the audio is in either state.

While this process may slightly affect true 5.1 channel programming, it is not possible to “trick” the system. Other methods that use level detection for determining whether programming is two channel or 5.1 can *always* be tricked by certain programming. It is impossible to reliably determine the true number of channels simply by the detecting the absence of Center, LFE, and/or the surround channels.

However, combining the above process of “partial upmixing” with level detection on the remaining discrete channels and careful timing of crossfades create an optimal process that cannot be fooled and will not affect discrete 5.1 channel audio. The AutoMAX-II mode is available on the LfRf upmixer and its operation is described in Chapter 4.

- NOTES -

Chapter 8: Specifications

Table 8-1 Electrical Specifications

Sampling Rate	48 kHz ($\pm 0.1\%$)
Output Delay	Main Out UPMAX Algorithm: 45 msec (fixed, all settings) Main Out UPMAX-II: 60 msec (fixed, all settings)
Upmixing Algorithms	Linear Acoustic UPMAX and UPMAX-II
Audio Word Length	24-bits
Digital Audio In	Four unbalanced female BNC connectors, comply with AES-3ID-2001/SMPTE 276M. Internal 75-Ohm termination.
Digital Audio Out	Four unbalanced female BNC connectors that comply with AES-3ID-2001/SMPTE 276M specifications.
Metadata Input/Output	RS-485, 9-pin female D-connector on rear panel
GPIO Port	TTL level, 25-pin female D-connector
Ethernet Port	RJ-45 female jack connector
Analog Monitor Output	
Frequency Response	20 Hz–20 kHz, ± 0.5 dB
Distortion	Less than 0.01% at 1 kHz Less than 0.02%, 20 Hz–20 kHz
Dynamic Range	Greater than 85 dB
Output Level	+4dBu @-20dBFS, +24dBu into 600 Ohms @0dBFS
SDI	
HD/SD-SDI I/O	Audio can be de-embedded (demuxed) from any of the four groups and re-embedded (muxed) into any of four the groups. Signals per SMPTE 299M-2004/292M-2004. Metadata can be extracted from VANC per SMPTE 2020 A or B. Supports 1080i/720P 50/59.94

Table 8-2 Mechanical Specifications

Dimensions	3.5 × 19 × 24 in (8.9 × 48.3 × 43.2 cm)
Net Weight	25 lb (11.3 kg) approx.
Shipping Weight	28 lb (12.6 kg) approx.
Power Requirements	90-240 VAC, auto-sensing, 50–60 Hz
Power Consumption	40 W maximum

Table 8-3 Environmental Specifications

Operating Temperature	0°C to 50°C, fan cooled
Non Operating Temperature (Storage)	-20°C to +70°C
Humidity	Up to 98% relative humidity, non-condensing
EMC Radiation Limits	FCC Part 15 Class A, ICES-003

Metadata Input Port

9-pin female D-connector with full-duplex RS-485 protocol running at 115 kbps. Pinout is compatible with SMPTE 207M. Pin-for-pin compatible with Dolby metadata sources (i.e. straight-through cable should be used).

Table 8-4 Metadata I/O Port Pinout

Pin	Connection
1	Shield
2	TX A asynchronous data out -
3	RX B asynchronous data in +
4	Ground
5	NC
6	Ground
7	TX B asynchronous data out +
8	RX A asynchronous data in -
9	Shield

Ethernet Port

Standard RJ-45 female connector that supports 100 or 1000BASE-T.

GPI/O Parallel Control Port

TTL level controls, active Low. GPI A and B are held low to activate their assigned function, while GPI 1-6 require momentary contact to ground to activate their functions. The 5-V output for external GPO indicators is limited by a self-resetting fuse.

Table 8-5 GPI/O Parallel Control Port Pinout

Pin	Function	Pin	Function
1	+ 5 V (150 mA)	14	+ 5 V (150 mA)
2	GPI A (Hold Low = On)	15	Ground
3	GPI B (Hold Low = On)	16	Ground
4	GPI C (Hold Low = On)	17	Ground
5	GPI D (Hold Low = On)	18	Ground
6	RESERVED	19	Ground
7	RESERVED	20	Ground
8	RESERVED	21	Ground
9	RESERVED	22	Ground
10	Unit Status: Open = OK	23	Unit Status: Common
11	Unit Status: Closed = OK	24	GPO 1
12	GPO 2	25	Ground
13	Ground		

Analog Output Port

DB-25 Female connector with an electrical pinout that matches the Tascam DA-88 format. Signals are +4dBu nominal with a digital input signal of -20dBFS, and have a maximum balanced output of +24dBu into 600 ohms.

Table 8-6 Analog Output Connector Pinout

Pin	Function	Pin	Function
1	Ch 8 Ro/Rt + Out	14	Ch 8 Ro/Rt - Out
2	Ch 8 Ro/Rt Gnd	15	Ch 7 Lo/Lt + Out
3	Ch 7 Lo/Lt - Out	16	Ch 7 Lo/Lt Gnd
4	Ch 6 Rs + Out	17	Ch 6 RS - Out
5	Ch 6 Rs Gnd	18	Ch 5 Ls + Out
6	Ch 5 Ls - Out	19	Ch 5 Ls Gnd
7	Ch 4 Subwoofer + Out	20	Ch 4 Subwoofer - Out
8	Ch 4 Subwoofer Gnd	21	Ch 3 Center + Out
9	Ch 3 Center - Out	22	Ch 3 Center Gnd
10	Ch 2 Right + Out	23	Ch 2 Right - Out
11	Ch 2 Right Gnd	24	Ch 1 Left + Out
12	Ch 1 Left - Out	25	Ch 1 Left Gnd
13	NC		

Table 8-7

Remote Volume/Mute/Return to Ref Port

Female DB-9 connector accepts TTL level controls, active Low. GPIs require a momentary contact closure to activate their assigned function. The 5-V output for external GPO indicators is limited by a self-resetting fuse.

Table 8-8 Remote Volume Port

Pin	Connection
1	Mute (GPI A)
2	Return to Ref (GPI B)
3	GPO A (High when Muted)
4	GPO B (High when at Ref)
5	Ground
6	Encoder Phase A Input
7	NC
8	Encoder Phase B Input
9	+5V, 200 mA Max