



AERO.one

**DTV Audio/Loudness Manager
Model LA-5100 V3 (5.1 + 2 channel)**

User Guide





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User Guide

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Chapter 1: Introduction

The Linear Acoustic AERO.one™ is a compact transmission loudness controller and up-mixer. The V3 model has the following features:

- AEROMAX® 5.1 and 2.0 channel loudness control engines
- Dual UPMAX® world-proven upmixing engines with AutoMAX-II™ auto upmixing for both main and secondary program audio inputs
- Secondary or “Local” +2 input for processing alternate program or separate local and network audio
- Dedicated full-time LtRt or LoRo stereo downmix output
- Dedicated EAS input
- GPI/O control of upmixing and EAS insertion functions
- Industry-leading metadata functionality for optimized processing
- Relay bypass of audio, SDI, and metadata connections for fail-safe operation
- HD/SD-SDI I/O with access to all 16 channels plus VANC metadata
- Dual redundant power supplies
- Optional dual Dolby® Digital (AC-3)/Plus (E-AC-3) encoder for 5.1+2.0 audio

1.1 Principles of Operation

The AERO.one is a simplified version of the Linear Acoustic AERO.air series of transmission loudness managers. It contains the same integrating long-term loudness controller, multiband short-term loudness controllers, UPMAX upmixing engine, and automatic, metadata, and GPI control

Specific processing presets and adjustments are discussed in Chapter 5.

Figure 1-1 shows the internal audio path of the AERO.one. It should be noted that this diagram is a general representation of signal flow and shows the optional Dolby Digital (AC-3) encoder. Please consult the appropriate section of this manual for specific functionality.

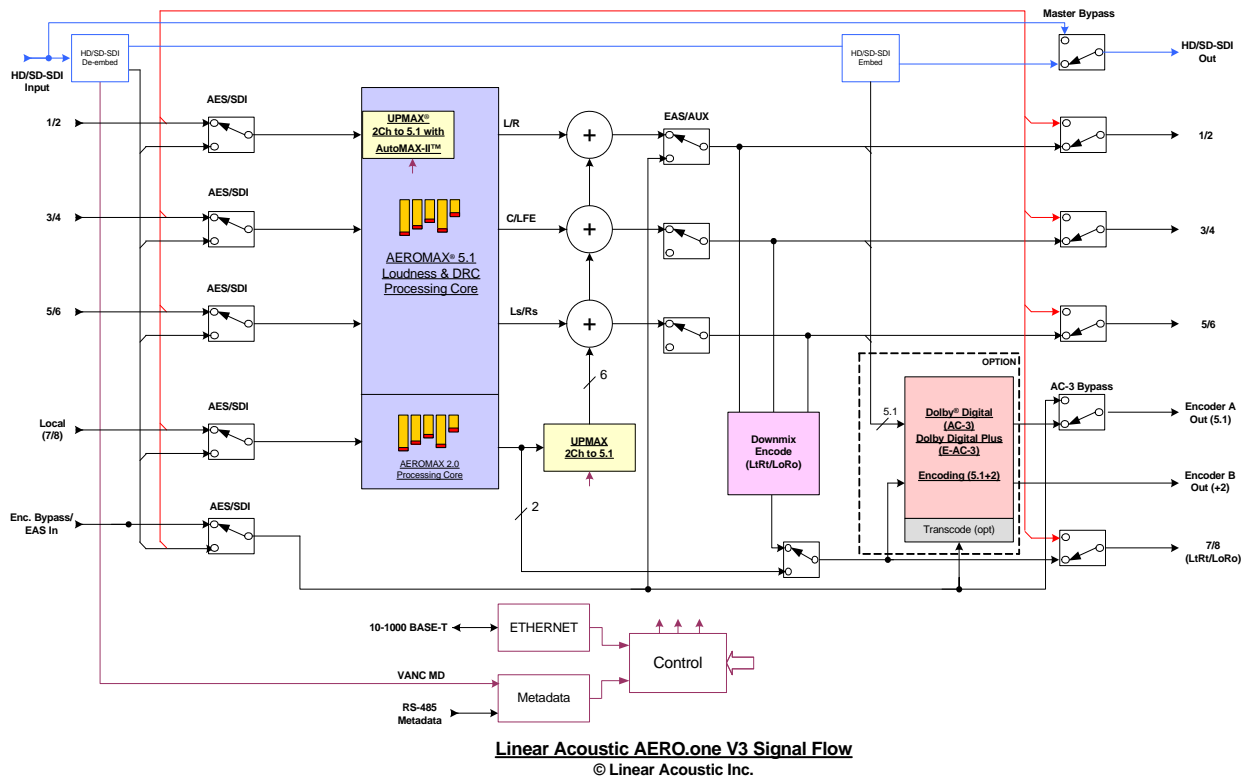


Figure 1-1 Audio Block Diagram

1.2 Location of AERO.one

In the signal flow of a modern television station, audio processors can be placed in a variety of locations depending on how the transmission path is designed. Our suggestion is to try to keep AERO.one near the final emission DTV encoder (i.e. Dolby Digital (AC-3) encoder) as the units work closely together and will benefit from short cable runs and common clocking.

1.3 Reference Levels

The AERO.one 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 setup and configuration information.

Chapter 2: Connections and Quick Setup

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

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:
 - Quick-start sheet to get you up and running
 - IEC power cords (style matches country of order);
 - this manual, and a handy black pen.
- 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.one installation requires:

- One standard rack space unit with ADEQUATE VENTILATION (the unit relies on convection cooling from side-panel vents);
- 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 reference. The unit can be synchronized to AES Input 1, AES Input 4, SDI, or Internal 48kHz. All inputs have SRCs but a master reference is required. Proper reference signal selection and application is imperative for an artifact-free installation. Clicks or pops can usually be traced back to improper reference configuration.

2.3 Rear Panel

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

2.3.1 Connection Ports

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

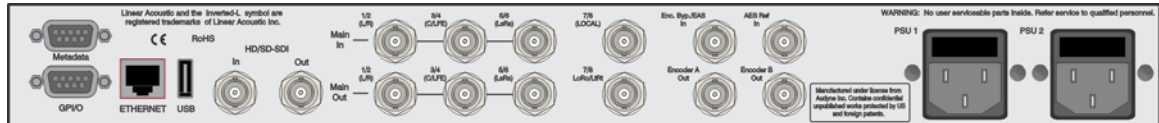


Figure 2-1 Rear Panel

- **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/SDI Output:** Option allows for de-embedding and re-embedding any of the 16 available channels in an applied HD or SD-SDI signal. VANC metadata extraction per SMPTE 2020 methods A and B is also supported.
- **GPI/O:** Connect dry contact closures here to control upmixing and EAS on/off. Note that GPI functions require held closures for the duration of the desired function. Status of upmixing will be reflected on the corresponding GPO pin.
- **ETHERNET:** Used for firmware upgrades.
- **Main Audio Input:** 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=Secondary +2 or Local program audio.
- **Enc. A Bypass/EAS Input:** Bypass input for Dolby Digital (AC-3) encoder A. Accepts output of external encoder for fail-safe operation. Also used for EAS input.
- **AES REf Input:** Apply 48kHz DARS (Digital Audio Reference Signal) here and choose AES Ref as clock source in Setup->I/O menu. Note that this is useful with some MPEG video encoders which provide an AES reference output.

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

- **Main Audio Outputs:** Main 48kHz processed digital audio outputs. 1/2 = Left Front/Right Front, 3/4 = Center/LFE, 5/6 = Left Surround/Right Surround.
- **LtRt/LoRo Output:** An LtRt or LoRo downmixed version of the main program is available from this output connector at all times. Main program audio is replaced with EAS (Aux) audio when commanded.
- **Encoder A Output:** AC-3 output of the 5.1 channel Dolby Digital encoder option
- **Encoder B Output:** AC-3 output of the 2.0 channel Dolby Digital encoder option

2.4 Quick Setup Notes

The Linear Acoustic AERO.one 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.

- Apply audio to AES Main Input 1/2 (or via SDI 1/2)
- In Setup -> I/O menu make sure Input 1/2 = AES 1/2 (or SDI 1/2)
- Set Clock Reference = AES 1 (or SDI)
- The front panel Reference indicator should be green.
- The main menu will default to showing input and output meters, and there should be activity on some or all of the first six meters.
- Processed audio will be available on the main outputs 1-6 and a two channel down-mixed version will be present on the 7/8 output.

The best way to learn the processor is to explore the different settings with audio applied and monitored. There are several 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. Modern broadcast television offers the ability to process less, and even the heaviest presets in the AERO.one are moderate by comparison to what might be found in traditional NTSC audio processors.

NOTE: *For best audio quality, the AERO.one should be the only processor in line with station audio. Please make sure to remove any legacy processing that might have been installed for NTSC operation as it will impair the performance capabilities of the AERO.one. We suggest checking drawings and physical installations to make sure that this old gear is removed.*

Chapter 3: Applications

The AERO.one is intended to be used in an emission (transmission) environment. This chapter discusses several applications, including describing how it can be used to support three common US terrestrial network configurations:

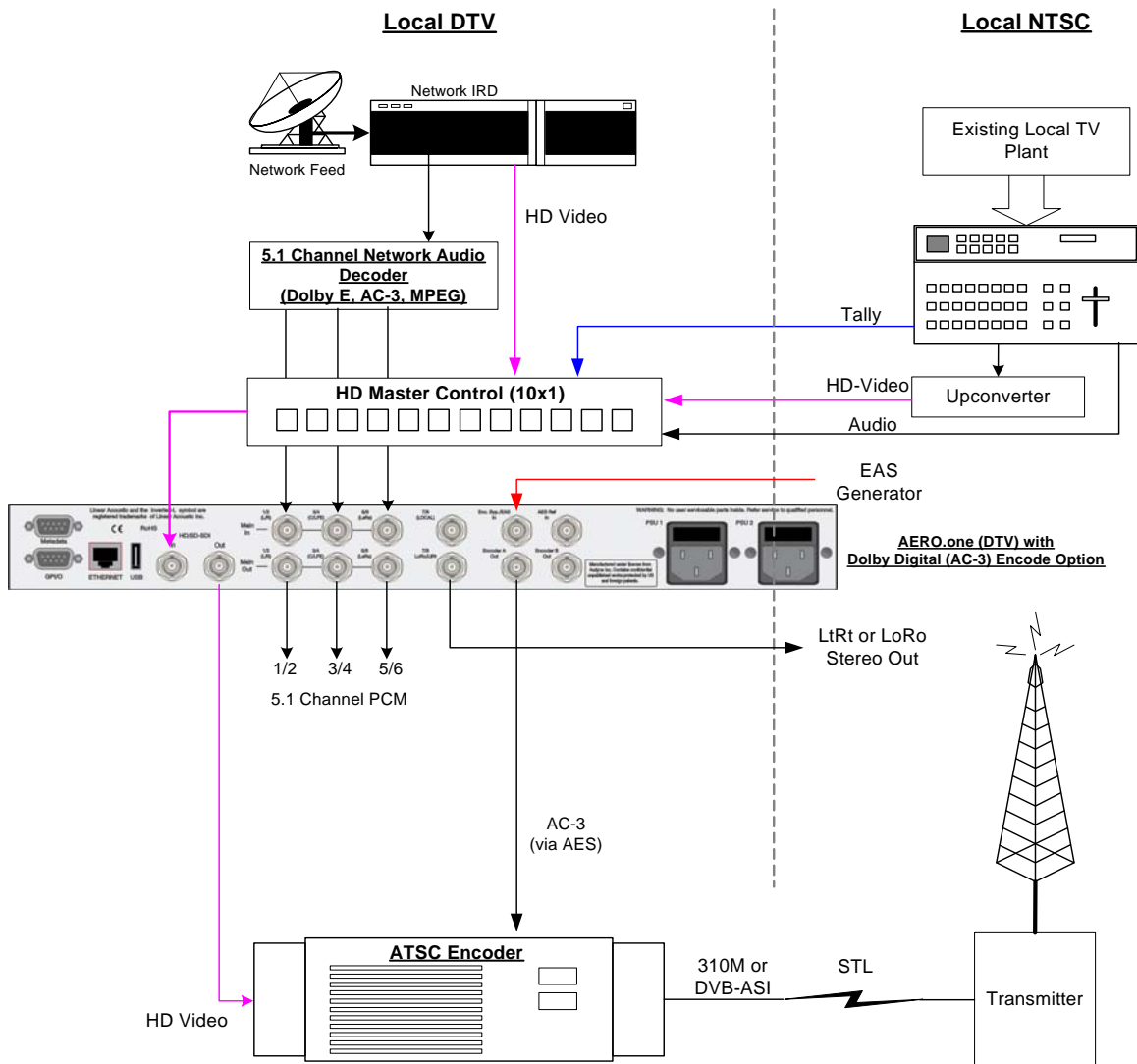
- AC-3 at 384/448 kbps (FOX, PBS)
- Baseband PCM (NBC, ABC after distribution via MPEG1 Layer 2)
- PCM/Dolby E within HD-SDI (CBS)

Additional information is provided that explores the challenges and requirements of audio metadata and how it can impact local television station audio setup. Specific descriptions for configuring local Dolby Digital (AC-3) encoders are also provided.

Although each environment has unique requirements, they are also very similar. The figures below show how the AERO.one can be integrated within each of the environments. First however, a general application diagram is shown below to illustrate how the unit might be applied. Facilities will differ, but the overall signal flow will be indicative of how the unit can be installed.

There are a few common feature that might be useful. For example, EAS audio can be applied to the AERO.one via a dedicated AES or SDI input. A stereo and matrix surround-compatible LoRo or LtRt signal is also generated from the main audio program and provided via a dedicated AES or SDI output.

AERO.one Typical Local Station Installation



- Features:**
- Automatic Loudness control & Upmixing
 - AES and HD/SD-SDI I/O
 - LtRt or LoRo Stereo digital out
 - Metadata fully supported but not required
 - GPIO control
 - Dual PSU
 - 5.1 channel Dolby Digital (AC-3) encoder option

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Figure 3-1 AERO.one general application

3.1 AC-3 Distribution System

Figure 3-1 shows how an AERO.one would be installed in a typical AC-3 distribution system that might be found at a PBS or FOX station. The Dolby Digital (AC-3) bitstream data rate is 384 or 448 kbps and is delivered within a standard 19.39 Mbps ATSC transport stream. This allows a local station to use a commonly available transport stream splicer to insert locally generated digital television video and audio streams. Alternately, a station could decode the network audio and video, insert local programming, then re-encode the audio and video for transmission to consumers (resulting in potentially detrimental side effects on the audio due to the loss of metadata).

As there is no way to deliver the professional audio metadata in this scenario, it is simply ignored and default values are set in the Dolby Digital encoder for audio coding mode, Dialogue Level (dialnorm), and Line and RF Mode DRC presets. The audio is then processed by the AERO.one so that it matches these metadata defaults. See section 3.3 below for a description of exactly how to set these parameters.

A built-in Dolby manufactured Dolby Digital (AC-3) encoder will be available as an option for the AERO.one in the near future. This option will simplify the wiring and configuration necessary when using an external encoder. It also adds the reliability of a redundant power supply and the flexibility of HD and SD-SDI I/O. Please contact your salesperson or the factory for availability.

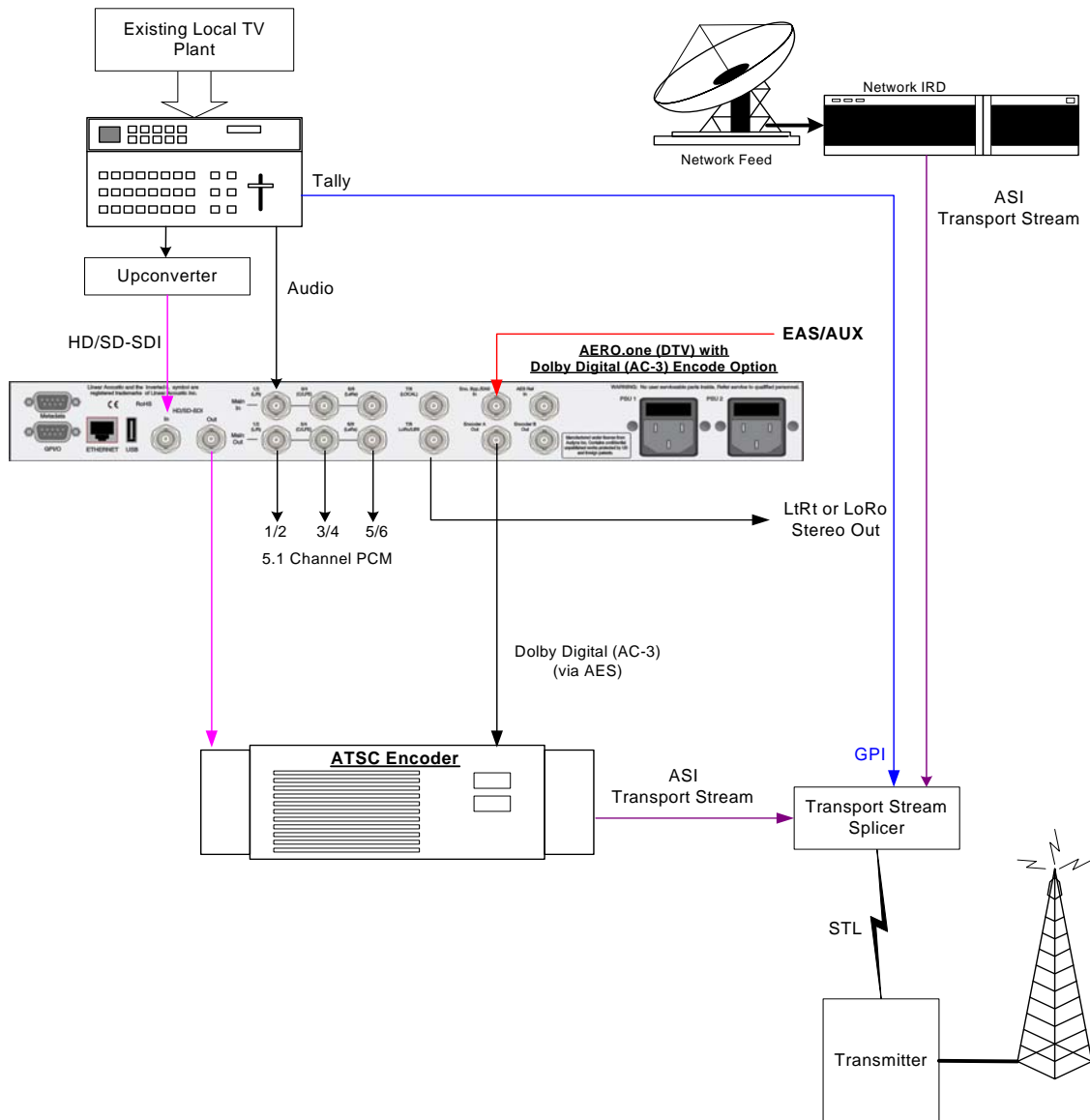


Figure 3-2 AERO.one within a typical AC-3 distribution system (FOX, PBS)

3.2 PCM Distribution System

Figure 3-2 shows how an AERO.one would be installed in a typical PCM distribution system. Note that audio coding will very likely be part of the satellite distribution gear. In the case of NBC, MPEG 1 Layer 2 is built in to the satellite encoder at the network side, and into the IRD (Integrated Receiver Decoder) units at each affiliate station. The system accepts PCM prior to distribution and produces PCM after reception, so it is considered a PCM distribution system.

Note that metadata is carried via the Data B channel of the IRD in the case of NBC. It is standard audio metadata and can be applied to the AERO.one directly as shown.

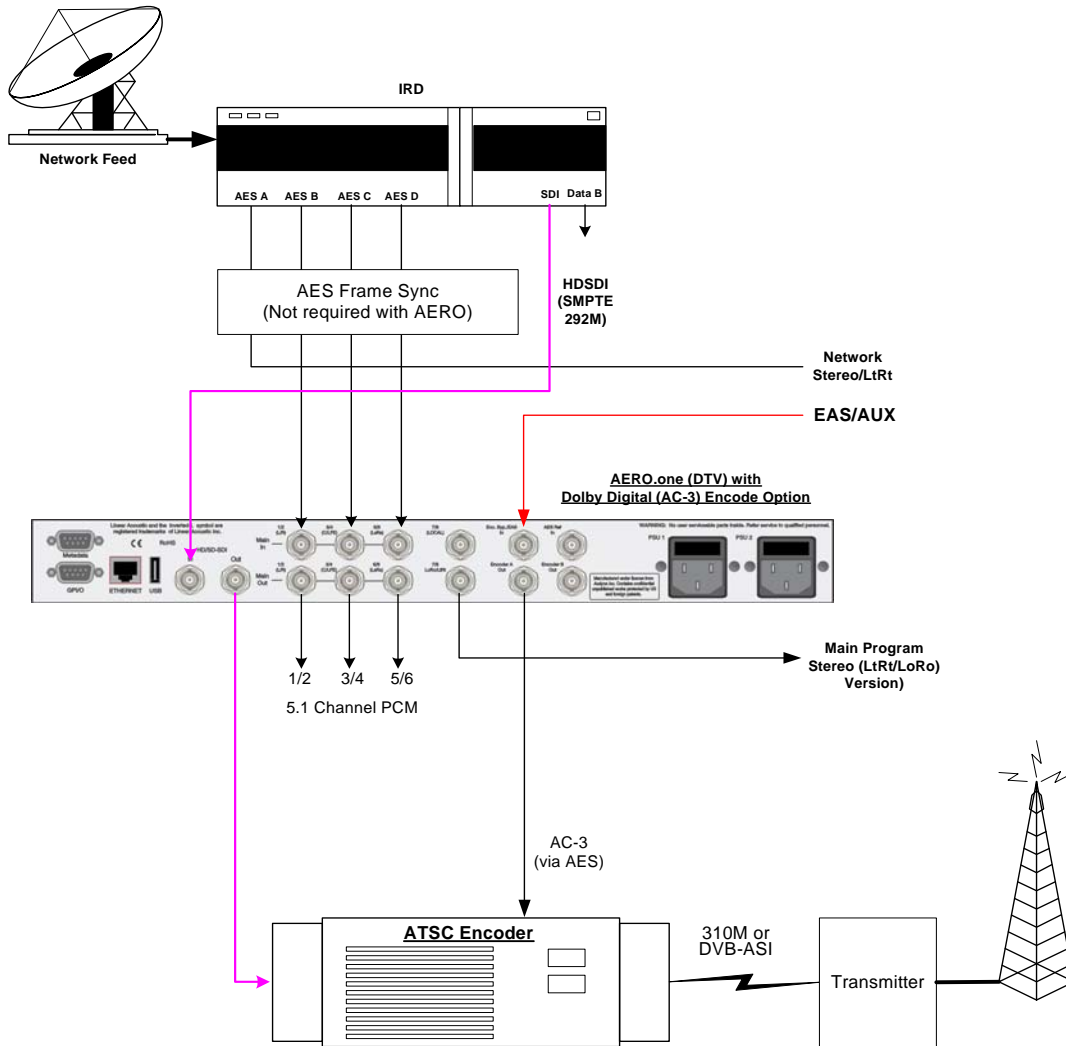


Figure 3-3 AERO.one in a typical baseband PCM system (i.e. ABC, NBC).

A built-in Dolby manufactured Dolby Digital (AC-3) encoder will be available as an option for the AERO.one in the near future. This option will simplify the wiring and configuration necessary when using an external encoder. It also adds the reliability of a redundant power supply and the flexibility of HD and SD-SDI I/O. Please contact your salesperson or the factory for availability.

NOTE: Exact IRD used may provide SDI-only connectivity and may not require frame synchronization. The AERO.one can be used in either scenario.

3.3 HD-SDI Distribution System

Figure 3-3 shows how an AERO.one would be installed in a typical HD-SDI environment, similar to that in use by the CBS and CW networks. In this case, audio and metadata are decoded and then embedded into an HD-SDI signal which can then be routed and switched through the plant and finally applied to the AEROMAX unit. Audio and metadata are de-embedded, processed, and Dolby Digital (AC-3) encoded for transmission.

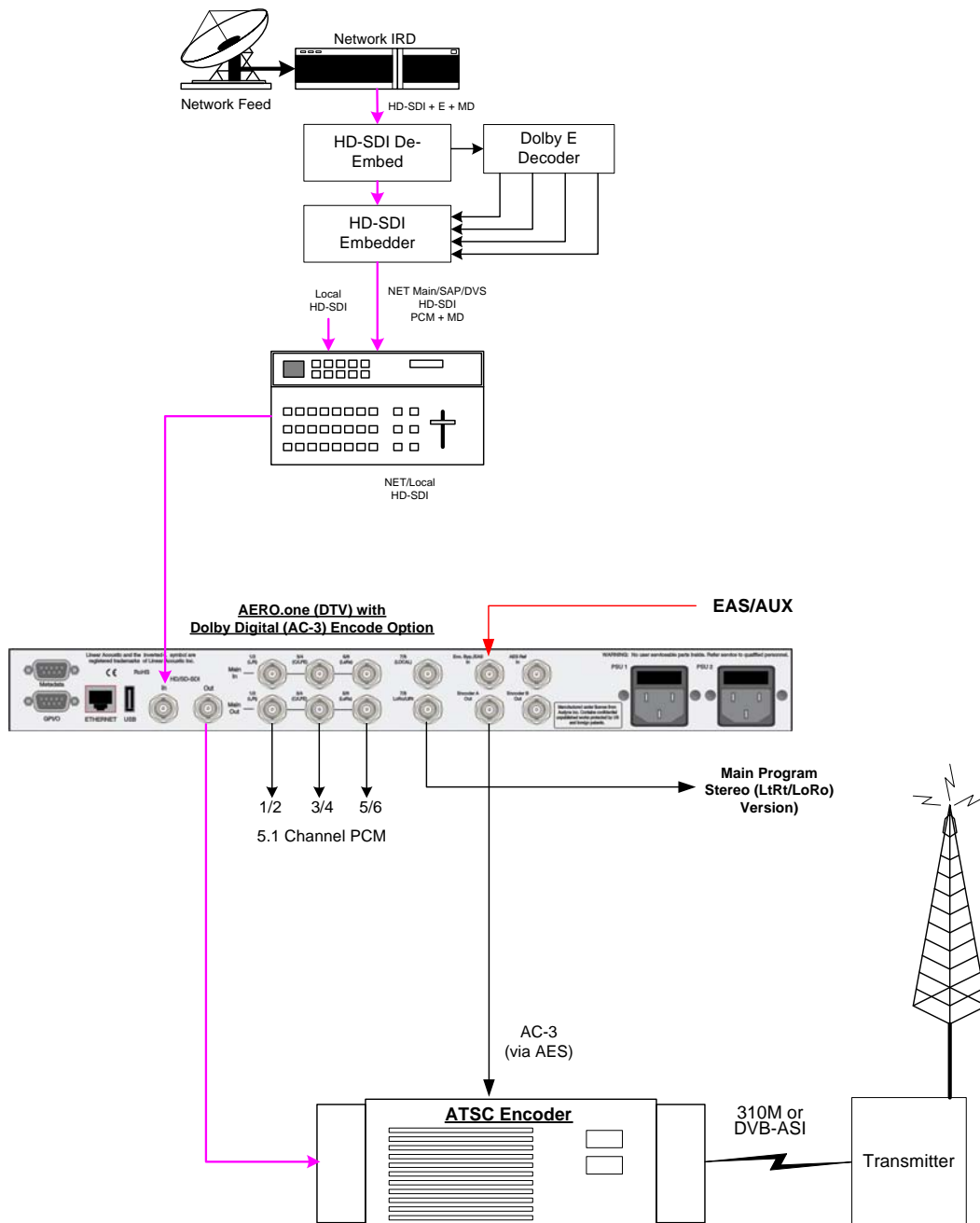


Figure 3-4 AERO.one in a typical HD/SD-SDI environment (CBS and others)

All three scenarios show how simply the AERO.one integrates into a television plant.

Additional features to support SAP and DVS or alternate channels of audio are present in the AERO.air (5.1) version of the product. Please contact your salesperson or the factory for more information.

3.4 Audio Metadata

Audio metadata, or data that describes the audio data, is used in the AC-3 system to represent important parameters such as the number of channels that have been encoded (audio coding mode or acmod), the loudness of the program with respect to dialogue (dialogue level or dialnorm), and dynamic range control (Line Mode and RF Mode DRC presets). Applied to an AERO.one, audio metadata can be used to control functions such as when to apply upmixing and eventually guide processing preset selection. Use of an AERO.one with metadata can also protect stations from metadata failure.

3.5 Dolby Digital Encoder Setup (external metadata)

In environments where the television network is providing reliable metadata to affiliate stations and the stations are able to access it, the AERO.one can work in tandem with this data to ensure audio of the highest reliability and consistency. When installing an AERO.one, network audio metadata should be connected only to the AERO.one.

By analyzing the incoming metadata, features such as upmixing can be automatically invoked when appropriate. The AERO.one is also capable of detecting metadata failure and responding appropriately. For example, if metadata were to fail during a 5.1 channel program, upmixing should not be invoked, nor should an AC-3 encoder be switched to 2/0 stereo mode, and both of these situations are prevented.

As the dynamic range, and hence the average loudness of the programming is being controlled by the AERO.one, metadata such as dialnorm and DRC presets should be set to fixed values. Suggestions are to set dialnorm to -24 and select Film Standard for both the Line Mode and RF Mode DRC presets. In this manner, the AERO.one will automatically pre-adjust the audio to match these settings.

3.6 Dolby Digital Encoder Setup (local metadata)

To use the AERO.one with a Dolby Digital encoder in an environment without external metadata, such as the type that might be delivered by a network, three critical values must be set in the AC-3 encoder:

- **Audio Coding Mode:** sets the number of channels to encode. Format is Front Channels/Rear Channels: 3/2 is for 5 channel, 2/0 is for stereo. LFE is enabled or disabled elsewhere to create 3/2L for 5.1.

- **Dialogue Level:** sets the dialogue loudness anchor (also known as *dialnorm*). Values vary between -31 and -1 and are correctly set with an ITU loudness meter such as a Linear Acoustic LAMBDA or LQ-1000 or equivalent.
- **DRC:** built-in dynamic range control system (also known as Line Mode (*dynrng*) and RF Mode (*compr*), driven by presets such as “Film Standard”.

Note that as they are intimately connected, you must set the dialogue level properly in order for the DRC (dynamic range control) system to perform as intended.

3.6.1 Audio Coding Mode

Also known as *acmod*, this value is shown, for example as 3/2L. This means three front channels, two rear channels, and LFE (i.e. the channel that feeds the sub woofer). Without metadata, it should be set to 3/2 and the LFE turned on so that the upmixing included with the AERO.one can be relied upon to keep all programming in 5.1 channel surround sound.

3.6.2 Dialogue Level

Properly setting the Dialogue Level parameter ensures that the dialogue volume heard by the consumer (or average level in programs without dialogue) is consistent between programs. Contrary to popular belief, there is no one “correct” value for unprocessed audio. As the value indicates the average dialogue loudness of a program and different programs will very likely have different loudness, they will also then have different dialogue level values. Correct values are found by measuring each program.

This section assumes no metadata is available and that the AERO.one is relied upon to match levels between programs. Only in this scenario is it then appropriate to set a single dialogue level value on the AC-3 encoder as the AERO.one is doing the job of matching each program to a single loudness value.

To set Dialogue Level:

1. Ideally, using either a Linear Acoustic LAMBDA or LQ-1000, or a Dolby LM100 will make this very easy because it displays the actual metadata from the encoded stream and the measured dialnorm value in real time, and the encoder can be quickly adjusted via its front panel.
2. Note that as long as the dialnorm metadata value and the measured audio match on average, setup is correct regardless of the actual final value of dialnorm. By “on average” we mean that over time the values should match more than they do not match, and some movement is OK (these are the good dynamics of the program making it through). The only audio that will produce a non-varying result is test tone (or horrendously over-aggressive processing) best to avoid both.

3.6.3 Dynamic Range Control

Dynamic range control profiles can be selected in the Dolby Digital (AC-3) encoder.

NOTE: *Dialogue Level must be set correctly for the DRC profiles to operate correctly.*

When using an AERO.one preset such as TV 5B Gen or Light (Television 5-Band General or Light), we suggest using is the Film Standard or Film Light DRC profile in the Dolby Digital encoder for both Line Mode and RF Mode. This allows consumers to restore some of the dynamic range. The AERO.one will still react to badly mismatched loudness levels, but will do so in combination with the DRC profile in the Dolby Digital encoder.

We strongly recommend using the minimum degree of dynamic range processing required. TV 5B Gen should provide more than enough pre-processing, and TV 5B Light might be better suited for some material. In this way, the system can balance between permanent changes to the input audio and metadata-based changes generated by the internal Dolby Digital (AC-3) encoder. Of course for proper operation this method requires programming that has been correctly produced.

AC-3 Encoder Metadata Value List

Specific audio metadata parameters that should be set in the AC-3 encoder are:

- Data rate: 384kbps minimum, 448kbps preferred for 5.1 channels
- Dialogue Level: -24 (with the AERO.one in TV 5B Gen)
- Channel Mode (acmod): 3/2L
- Line Mode Profile: Film Standard or Film Light
- RF Mode Profile: Film Standard or Film Light

Other parameters are set to default values and will be adjustable in the future.

General Dolby Digital (AC-3) Encoder Setup

Note that when feeding video encoders such as those from Tandberg, an AES reference signal is supplied *by the video encoder* specifically for feeding back to the AC-3 encoder. If the encoder requires this, as the Tandberg models do, it must be used. It seems that this special reference signal is derived from a 27MHz clock that is internal to the video encoder that may or may not be referenced to the plant. Failure to use this reference may cause PTS values to drift when observed on a transport stream monitor, and can cause audible glitches when buffer over or underflow results.

3.6.4 Audience Measurement Systems

Audience measurement systems such as the ones available from Nielsen (via Norpak) perform their functions by inserting a watermark into the audio signal. Careful testing by Nielsen and Linear Acoustic has shown that the watermark signal is not disturbed by any of the AERO factory presets or upmixing, and that locating the processor before or after the watermarking equipment has no impact on audience measurement reliability.

In order to minimize any chance of audibly unmasking the watermarking signal, Linear Acoustic and Nielsen recommend that where possible, watermarking equipment be installed *after* any audio processing equipment. However, audible unmasking is highly unlikely with standard presets. This arrangement is not possible when the Dolby Digital (AC-3) encoder is included in the processor, and so the watermarking equipment must be installed prior to the AERO unit. This is completely acceptable and in use by many stations.

3.6.5 Emergency Alert Systems (EAS)

As EAS systems rely on passing data modulated as audio through the audio channels of a broadcast facility, it is important to ensure that the path is capable of preserving this modulation. Ideally, EAS systems should be inserted after audio processing equipment and this is how the EAS Input of the AERO.one unit is configured. Main audio is muted and externally supplied EAS programming is inserted into the Left and Right output channels post processing.

As shown below, EAS mode is active when the top menu shows “EAS” to the left of the last two input meters. When not in EAS mode, this area will be blank, although the input meters will still show activity of applied audio.

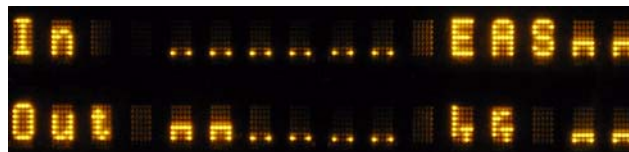


Figure 3-5 Main Menu showing EAS mode active

Chapter 4: Detailed Operation

This chapter discusses in detail the structure of the Linear Acoustic AERO.one, how to use the front panel interface to access the menus, accessing and saving factory and user processing presets, and GPI and Metadata setup.

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 the end of this chapter for a menu tree.

4.1 Main Menu (Signal Status Meters)

Audio I/O Meters

Shows the main audio I/O for the unit, plus the status of EAS functionality. Signal presence meters for each audio channel applied to the processing engines are followed by outputs from the processing and downmix engines. The left display shows audio present on the 1/2 input, 1/2 output, and the LtRt output. The right display shows EAS audio active and replacing the main audio, present on the 1/2 out and LtRt out.



Figure 4-1 Main Audio I/O Meters

Audio Input and Output Meters

The next two menus show output and upmixing status with channel labels. Note that upmixing is indicated as “D” (for Disabled), “U” for Upmix on, lower case “u” for controlled by GPI but not activated. This menu can also show when Upmix is set to Auto, and the same indications apply.

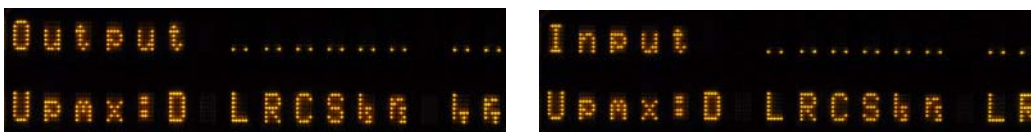


Figure 4-2 Audio signal presence and upmix indication

AGC Meters

Show the gain reduction status of the Input AGC and the Multiband AGC. More downward meter excursion indicates lower gain (more gain reduction). The center dot in each of the AGC bands shows 0dB at the center of the AGC range.



Figure 4-3 AGC Gain Reduction metering

GPI Status Display

Shows the current status of each of the four GP Inputs, plus the active preset. The left screen below shows GPI 1 set for EAS. A lower case “e” indicates that EAS is selected but the GPI is not active. The right screen below shows an upper case “E” indicating that EAS mode is active.



Figure 4-4 GPI Status Screens

Metadata Status (Pgm 1-8)

Shows the status of applied metadata signal (485 or VANC). The Program Config is shown along with the audio coding mode (acmod) and dialnorm for each of the possible eight programs.



Figure 4-5 Metadata Status Screen

4.2 Statistics Menu

Provides details about software, firmware and DSP versions, DSP communications status, as well as listing any installed options. This menu also shows the presence for each of the AES pairs, 1-8.

- **Firmware Version:** 5100.04.05
- **Device Options:** SDI
- **App Uptime:** Shows time running in d:hh:mm:ss
- **Ref Chg:** Counts the number of times the system reference has changed since reboot or reset
- **DSP 1 Version:** V1.5.0
- **DSP 1 TX Errors:** Shows transmission errors between micro and DSP1.
- **DSP 1 Rx Errors:** Shows receive errors between micro and DSP
- **DSP 2 Version:** V2.6.0
- **DSP 2 TX Errors:** Shows transmission errors between micro and DSP1.
- **DSP 2 Rx Errors:** Shows receive errors between micro and DSP
- **AES In Detect:** Shows presence of AES carrier on any of the 8 pairs (only four pairs are currently used to support 8 audio channels). See below for a picture of the screen



Figure 4-6 AES presence detect for applied pairs (1-4 are used)

4.3 Setup Menu

Entering the Setup menu gives access to **Presets/Processing adjustment**, **I/O settings**, **Communication** settings and **System** settings. With the exception of Presets/Processing menu which is described in detail in Chapter 7, all other settings will be described below.

4.4 Presets - See Chapter 5

4.5 I/O Menu

The I/O menu is where the input and output audio, control, and metadata signals are selected to interface with the loudness control and upmixing engines.

4.5.1 Master Bypass

Enabled de-energizes the audio, metadata, and SDI relays for a hard bypass.

4.5.2 GPI Control (Flashing “g”)

When enabled the unit is under GPI control and most changes are not allowed. This will be indicated in the menus by a lower-case flashing “g” in the left corner of the display.

4.5.3 GPI Function (GPI 1-4)

Selects the function of each GPI input. Selections are:

- **None:** No function for this GPI (as the name would lead you to believe)
- **Pri Preset:** GPI selects the preset defined by GPI Preset (4.5.4 below) for the Primary Program (i.e. input channels 1-6).
- **Sec Preset:** GPI selects the preset defined by GPI Preset (4.5.4 below) for the Secondary or Local input channels (7/8).
- **Upmix Off/On:** Enables upmixing for Left/Right front applied to Main In 1.
- **Upmix On/Off:** Reverse polarity of above.
- **Upmix Off/Auto:** Enables AutoMAX automatic upmixing to be enabled (see below for a description of AutoMAX).
- **Upmix Auto/Off:** Reverse polarity of above.
- **Full EAS Off/On:** 9/10 EAS replaces primary and secondary programs.
- **Pri EAS Off/On:** 9/10 EAS replaces primary program.
- **Sec EAS Off/On:** 9/10 EAS replaces secondary program.
- **Encod In Off/On:** Enables Transcoder input (option).

- **Encod In On/Off:** Reverse polarity of above.
- **DMX -> 7/8 Off/On:** Sends downmix of main program audio to 7/8 output.
- **DMX -> 7/8 On/Off:** Reverse polarity of above.
- **Local Off/Replace:** Replaces main program audio with audio from Secondary program on 7/8 input. This audio has its own processing core (Secondary Preset) and its own upmixer, both of which can be set the same or differently than the Main program. There is no auto mode as the Local input can only be two channels.
- **Local Replace/Off:** Reverse polarity of above. Note that selecting this will engage Local audio until a contact closure is applied to stop it.

4.5.4 GPI Preset (GPI 1-4)

Defines which of the stored presets a GPI will recall. Any of the sixteen stored presets are available for selection.

4.5.5 Upmixing

Upmixing is the process whereby two channel audio can be transformed into 5.1 channels of audio resulting in a pleasing and engaging manner. Dialogue remains centered, and music and effects are present in the front and rear channel in proportion to their original phase and magnitude. Original production values are maintained and the chances of downstream re-processing (by consumer gear) are virtually eliminated.

The industry standard Linear Acoustic UPMAX[®] algorithm is incorporated in the AERO.one. This allows for realistic 5.1 channel audio to be created from normal stereo or LtRt inputs. Two channel mono can also be accommodated in an audibly pleasing manner.

Critically, the UPMAX algorithm is also 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.

AutoMAX-II[™] Automatic Upmixing

AutoMAX is a method for automatically controlling the upmixing function of the AERO.air 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 up-mixed 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.

4.5.6 Upmix Control

Manual control of upmixer is as follows (note that if under GPI control, a flashing “g” will be displayed and no adjustments will be permitted):

- **Set On Now:** Enables upmixing for Left/Right front applied to Main In 1/2.
- **Set Off Now:** Disables upmixing
- **Auto:** Enables auto detection mode for upmixing, see below for details

4.5.7 Upmixing Setup

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

- **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.
- **LCR Sum to LFE:** Creates a downmix compatible low pass filtered bass enhancement signal to feed the LFE channel.
- **LCR Sum Level:** Sets the level of the created bass enhancement signal sent to the LFE output. Default is 12%.
- **Upmix Dsc Thresh:** Sets the threshold for the AutoMAX-II algorithm to decide that the input audio is not 5.1. Audio present in the Center, Left Surround, or Right surround channels above this threshold will trigger the auto upmixer to start upmixing the LfRf (channels 1/2) audio. Default is -75dB.
- **Upmix to Discr:** Sets the crossfade speed for disabling the upmixer and returning to discrete. As there is no loss of any audio, speed can be set to taste without fear. Default is Medium and which is fast enough for all content we have tested.
- **Discr to Upmix:** Sets the crossfade speed for enabling the upmixer. Since upmixing needs to be enabled faster than it needs to be disabled, the default setting is Quick.

WARNING: It is always best to explicitly signal any upmixer when to upmix and when not to upmix. Typically this would be via metadata or a contact closure. The AutoMAX-II process is for situations where closures or metadata are not available.

4.5.8 Local Input Settings

- **Local Input** - Manual control of Local Input
 - **Set On Now:** Main program audio is replaced by Local Input audio
 - **Set Off Now:** Main program audio is NOT replaced by Local.
- **Loc Insert Type** - Controls upmixer for Local Input:
 - **Upmix:** Local audio is upmixed before replacing Main Program audio.
 - **2-Channel:** Local audio is passed as 2-channels, replacing only LfRf of Main.
- **Loc Upm Ctr Wdth** - Controls Center Width of Local upmix
 - **33%** = Default
- **Loc Upm Sur Dpth** - Controls Surround Depth of Local Upmix:
 - **100%** = Default
- **Loc LCR SumToLFE** - Controls LFE creation for Local upmixer
 - **Enable** = Default
- **Loc LCR Sum Lvl** - Controls level of LFE creation for Local upmixer
 - **12%** = Default

4.5.9 Channel 7/8 & Downmix Setup

Downmix Type

The AERO.one unit produces a full-time downmix of the main program audio on the channel 7/8 output. This downmix can be selected as stereo LoRo or surround compatible stereo LtRt. Note that due to the required phase encoding, selection of LtRt will increase the latency of the 7/8 output to 21 milliseconds.

Downmix -> 7/8

Manual control, selects downmix of Main Program as output for 7/8. This can also be controlled by GPI, please see above.

Output 7/8 Trim

Allows output level of Main Out 7/8 to be attenuated over the range of 0dB to -18dB.

4.5.10 Reference Setup

Setup Output Reference for the AERO.one. A stable, high quality, non-interrupted signal should be applied and chosen for reference. Seems silly to have to even write this, but remember this choice determines the ultimate quality of your on-air audio, possibly the stability of your multiplexer. Outputs of crash switches are not recommended.

Clock Src Type

If the chosen clock source is from a stable reference, choose stable. For instances where clock reference might switch, choose Changing to minimize (but not eliminate) audio corruption. We strongly recommend using a stable reference source!

Clock Source (Output Reference)

AES 1, AES 4, SDI, or Internal 48kHz can be selected as the reference for the AES audio outputs. Note that if a selection is made for a source that is not applied, the front panel Ref LED will be red, and the AERO.one will revert to internal 48kHz to maintain audio. All inputs have sample rate converters on them so asynchronous audio can be accepted as long as the reference is set correctly.

NOTE: *If SDI Embed is enabled, Clock Source will be forced to SDI for proper timing.*

4.5.11 Audio Input Sources (Channels 1-8)

Selects between applied AES audio or audio de-embedded from SDI. Each of the input pairs 1/2, 3/4, 5/6, and 7/8 can be sourced from its corresponding AES input pair or any SDI pair. Default is AES.

4.5.12 SDI Embed

Selects whether to enable audio re-embedding or pass the SDI signal through the AERO.one untouched. Note this does not affect de-embedding which is active at all times. See note above about Clock Reference when SDI Embed is enabled.

4.5.13 SDI Out Destination (Channels 1-16)

Selects source of audio for re-embedding into an applied SDI signal. Choices are Mute, AES (corresponding pair), and any of the SDI pairs. This menu can be used as a router or pair shuffler within SDI.

4.5.14 Metadata Setup

The AERO.one has been designed to take best advantage of metadata. The incoming Dolby or Dolby-compatible metadata stream is parsed on a program-by-program basis, and the dialog loudness and audio coding mode (channel count) parameters are used to scale processing and control upmixing respectively. Setup of metadata can be accomplished in the AERO Setup -> I/O menu.

AERO.one Metadata Setup Procedure:

- **Source: RS485/VANC-** Selects between the RS-485 serial metadata input and metadata extracted from the Vertical Ancillary (VANC) of an applied HD-SDI signal. For VANC metadata there are several other settings:
 - **VANC MD is Async:** Disabled for standard VANC metadata, Enabled for CBS-style VANC metadata. Default is Disabled.
 - **VANC MD DID:** Selects the Data ID of the metadata signal in hex. SMPTE RP2020 recommends this value be set to 0x45. Currently CBS network uses 0x50 but will be changing to 0x45. Default is 0x45.
 - **VANC MD SDID:** Selects the Secondary DID of the VANC metadata signal in hex. SMPTE RP2020 recommends this value be set to 0x01 and this is the default.
 - **VANC MD Line:** Selects the video line to extract VANC metadata from. SMPTE RP2020 recommends that metadata be inserted on Line 9, but we have seen a great deal of variation, thus the default is Auto.
 - **Pri MD Pgm Sel:** Selects the source for metadata for the Primary program. Default is Pgm 1 (i.e. in a 5.1+2, Pgm 1 is the 5.1).
 - **Sec MD Pgm Sel:** Selects the source for metadata for the Secondary program. Default is Pgm 2 (i.e. in a 5.1+2, Pgm 2 is the 2.0).
- **MD = 2/0 Pgm Sel:** Describes which consumer metadata program will be used to control the processing. Default and most common is Pgm 1 and this is the default.
- **MD = 2/0 Function:** Describes what will be controlled when the selected metadata program has an audio coding mode (acmod) of 2/0. Selections include:

- **Upmix LFRf:** Enables upmixing for Left/Right front applied to Main In 1.
- **Local Replace:** Enables Local input to replace Main Program audio.
- **Off**
- **System Dialnorm:** Specifies the “fallback” dialnorm value which will be used if meta-data is not applied or if it fails. Default is -24.
- **Pri Dialnorm** - If set to yes, then the incoming dialnorm parameter is applied to the incoming Primary Program audio to scale it prior to processing. Essentially, this allows the audio to be centered in the minimum activity area of the processing. If dialnorm was set correctly upstream, then the processing will be minimal. If dialnorm does not match the audio, the AERO.one will realign it in real time. Default is Do Not Apply.
- **Sec Dialnorm** - As above, but for the Secondary or Local program audio. Default is Do Not Apply.

4.6 AC-3 Encoder Menu (option)

The Dolby Digital encoder option provides two encoders, the first capable of encoding up to 5.1 channels and the second capable of up to two channels. Both Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) are supported.

- **AC-3 Bypass:** Enabling this option invokes a hard-relay bypass of the Enc. A Bypass Input to the Encoder A output.
- **Audio Bit:** Controls the audio/non-audio bit of the channel status bits present on Encoder A and Encoder B outputs. Default in Non-Audio data.
- **Validity Bit:** Controls the validity bit of the channel status bits present on Encoder A and Encoder B outputs. Default is Valid.
- **Encoder Mode EnA:** Sets the encoder type for Encoder A to DD for Dolby Digital (AC-3) or DD Plus for Dolby Digital Plus (E-AC-3). Normal default is DD.
- **DD+ Rate EnA:** Sets the output data rate for Encoder A when it is in Dolby Digital Plus mode. Default is 448 kbps.
- **DD Rate EnA:** Sets the output data rate for Encoder A when it is in Dolby Digital (AC-3) mode. Default is 448 kbps.
- **Acmod EnA:** Sets the acmod or Encoding Mode for Encoder A. Notation is front channels/rear channels/LFE. So 3/2L is 5.1 while 2/0 is 2.0 stereo.
- **Line Mode EnB:** Choose the Encoder B Dolby Digital DRC profile for Line Mode

dynamic range compression (i.e the default mode for the stereo line-level outputs of most set top boxes and TV sets). Choices include Film Std, Film Lgt, Music Std, Music Lgt, Speech, and “None”. Default is Film Std.

- **RF Mode EnB:** Choose the Encoder A Dolby Digital DRC profile for RF Mode dynamic range compression (i.e the default mode for the RF re-modulated outputs of most set top boxes). Choices include Film Std, Film Lgt, Music Std, Music Lgt, Speech, and “None”. Default is Film Std.

WARNING: *It is **Strongly** recommended **not** to choose “None” mode for either Line Mode or RF Mode. This is a test mode intended for controlled laboratory use and not for over the air applications. Choosing “None” mode may result in unnecessary protection limiting occurring during downmix, so “None” is not really None! Choose Film Lgt. for minimal DRC action.*

- **Dialnorm EnA:** Using an ITU loudness meter (Linear Acoustic LAMBDA, LQ-1000 or equivalent), set the transmitted dialnorm value here. Note that the factory default preset of TV 5B Gen is pre-adjusted to produce -24 on average with typical program audio inputs so the default setting is -24. This should be measured as part of initial system setup. The actual value does not matter as long as it matches the measurement.

NOTE: *To correctly set dialnorm, the loudness of the output audio must be verified using an ITU meter. For compliance, any value between -1 and -31 could be valid, as long as it matches the actual measured audio. Default is -24.*

- **Encoder Mode EnB:** Sets the encoder type for Encoder B to DD for Dolby Digital (AC-3) or DD Plus for Dolby Digital Plus (E-AC-3). Normal default is DD.
- **DD+ Rate EnB:** Sets the output data rate for Encoder B when it is in Dolby Digital Plus mode. Default is 448 kbps.
- **DD Rate EnB:** Sets the output data rate for Encoder B when it is in Dolby Digital (AC-3) mode. Default is 448 kbps.
- **Acmod EnB:** Sets the acmod or Encoding Mode for Encoder B. Notation is front channels/rear channels/LFE. So 2/0 is 2.0 stereo while 1/0 is mono.
- **Line Mode EnB:** Choose the Encoder B Dolby Digital DRC profile for Line Mode dynamic range compression (i.e the default mode for the stereo line-level outputs of most set top boxes and TV sets). Choices include Film Std, Film Lgt, Music Std, Music Lgt, Speech, and “None”. Default is Film Std.
- **RF Mode EnB:** Choose the Encoder B Dolby Digital DRC profile for RF Mode dy-

dynamic range compression (i.e the default mode for the RF re-modulated outputs of most set top boxes). Choices include Film Std, Film Lgt, Music Std, Music Lgt, Speech, and “None”. Default is Film Std.

WARNING: It is **Strongly recommended not** to choose “None” mode for either Line Mode or RF Mode. See Above.

- **Dialnorm EnB:** Using an ITU loudness meter (Linear Acoustic LAMBDA, LQ-1000 or equivalent), set the transmitted dialnorm value here. Note that the factory default preset of TV 5B Gen is pre-adjusted to produce -24 on average with typical program audio inputs so the default setting is -24. This should be measured as part of initial system setup. The actual value does not matter as long as it matches the measurement.

NOTE: To correctly set dialnorm, the loudness of the output audio must be verified using an ITU meter. For compliance, any value between -1 and -31 could be valid, as long as it matches the actual measured audio. Default is -24.

4.7 Communication Menu

Display and adjust the IP, Net Mask, Port addresses in this menu. The MAC address can also be found in this location.

4.8 System Menu

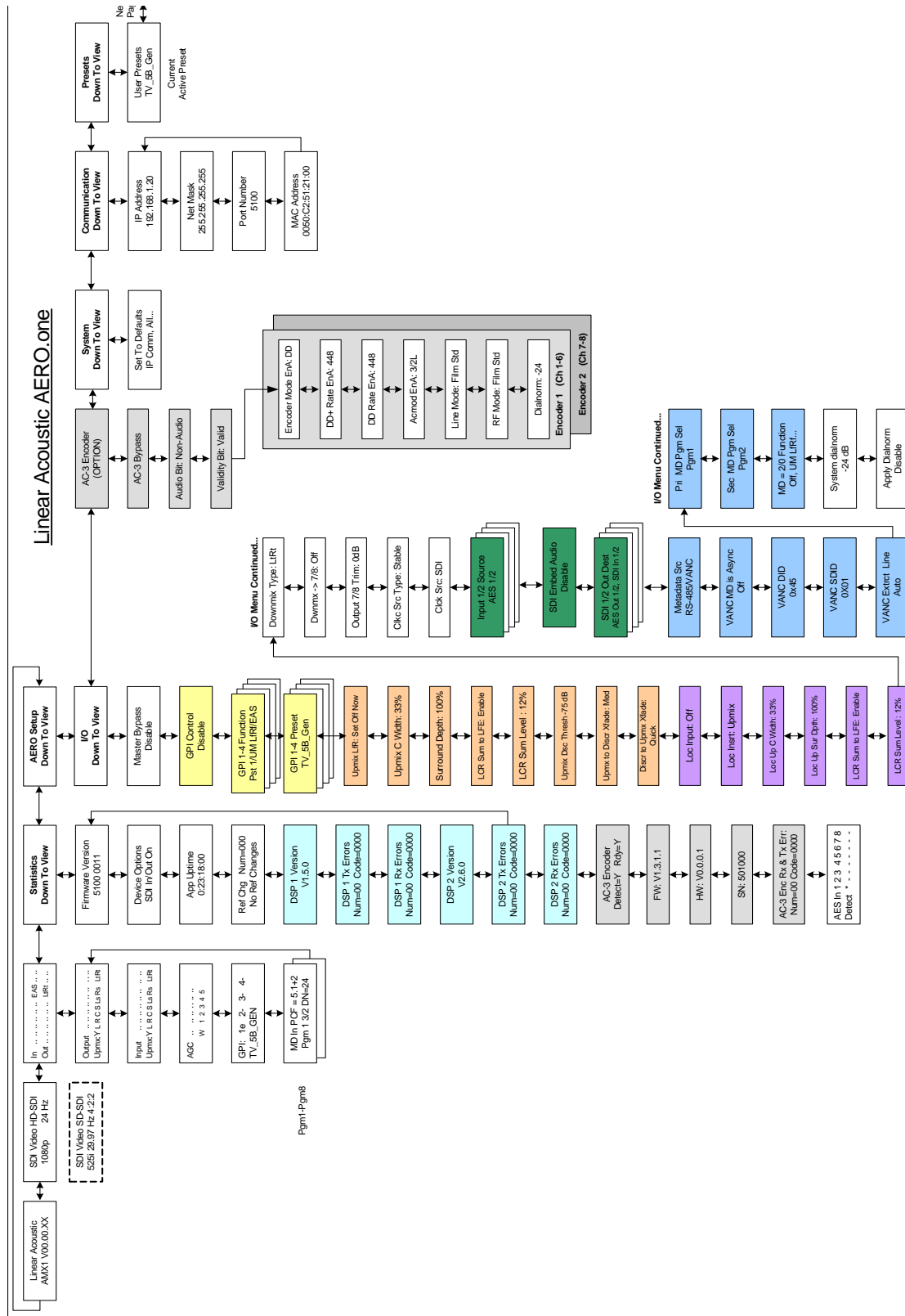
Allows reset to factory defaults. Note that IP settings, Presets, and I/O can be reset individually or “All” can be chosen to be reset simultaneously.

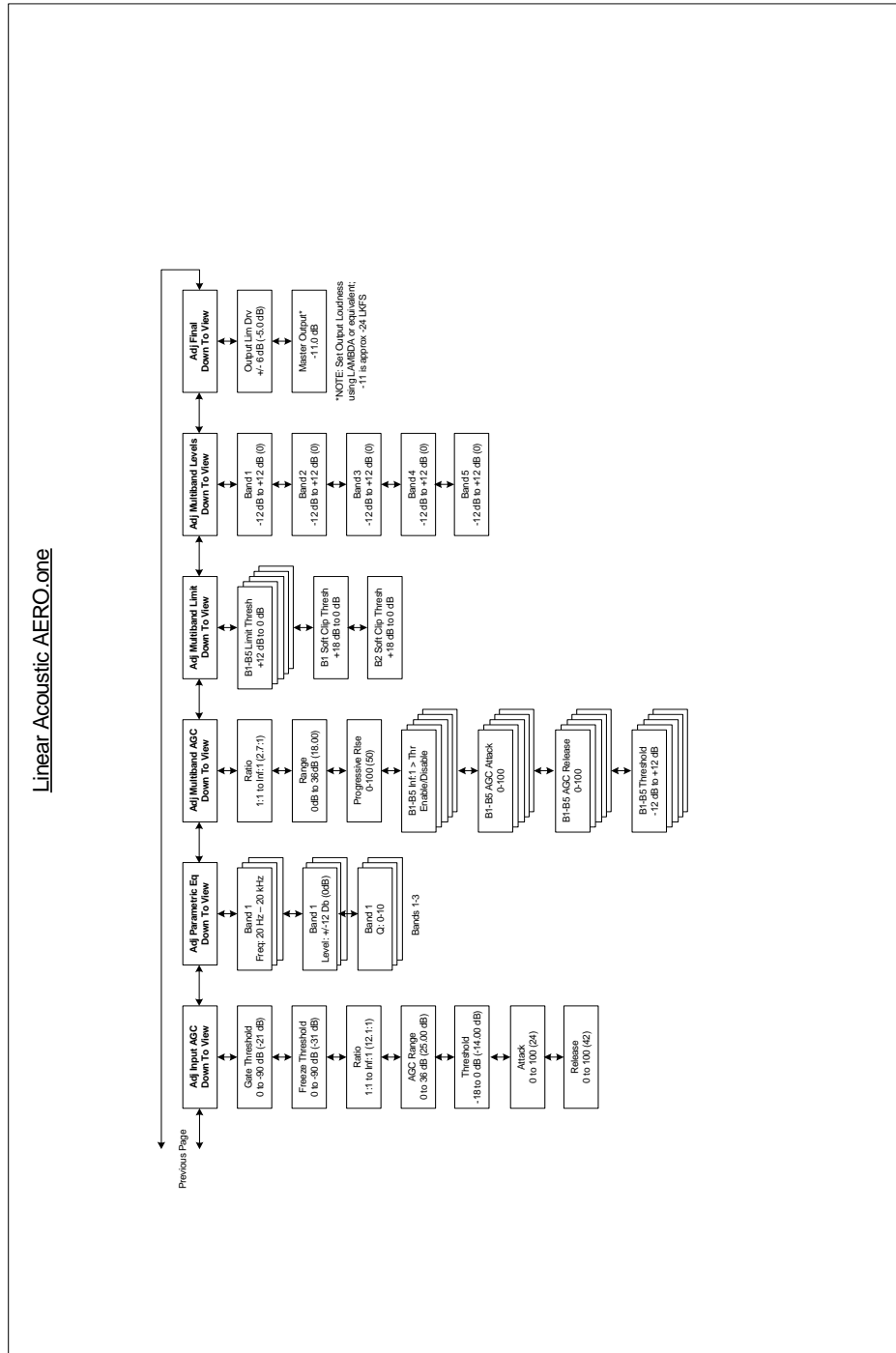
4.9 SDI Status Menu

Pressing the left arrow key when at the top menu will display the status of an applied SDI signal. The SDI input is auto-sensing between SD and HD, and although it is not recommended to hard switch input streams between SD and HD if a clean switch is desired, the unit will automatically track the change. HD or SD, frame rate and frequency are displayed. SD-SDI is shown on the left below, and HD-SDI is shown on the right.



Figure 4-7 SDI Status Screens





Chapter 5: Presets, Processing & Metadata

The Linear Acoustic AERO.one contains multiple factory-programmed processing presets that have been developed after many hours of listening and experimenting, using hundreds of program sources across all genres for tuning. Preset creation is an ongoing process, and we regularly implement new presets based on customer feedback.

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.

5.1 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.

WARNING: Factory presets CAN be over-written. Default values for these presets are given at the end of this chapter (just in case).

- **TV 5B GEN** - 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 -24 LKFS.
- **TV 5B 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. Lighter processing may result in more *momentary* variation from the loudness target.
- **TV 5B HVY** - 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 LOUD** - Similar to TV 5B HVY, but louder and more punchy.
- **MUSIC HVY** - Dense processing for TV music channel applications.
- **SPORTS** - Multiband attack and release times modified for a smooth and consistent result with typical network sports programming. Less range for both the Input AGC and Multiband AGC.

- **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** - Bypasses all processing except for the final output limiter which is set only to prevent overload.

5.1.1 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.

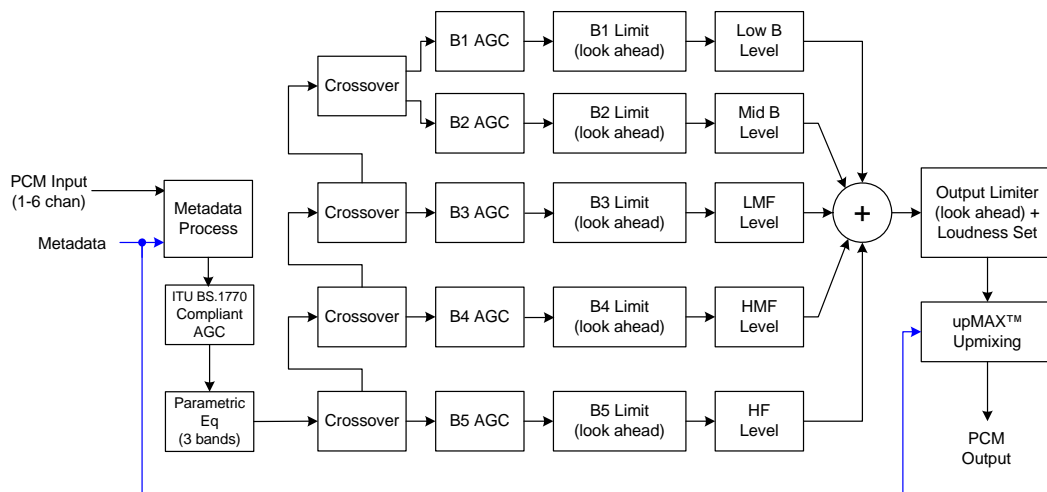


Figure 5-1 General signal flow of the AERO.one processing core

5.1.2 Adjusting Processing

Following is a description of all adjustable core processing parameters. Defaults are shown for the DTV 5B Gen preset, other presets will obviously vary some or all of these parameters.

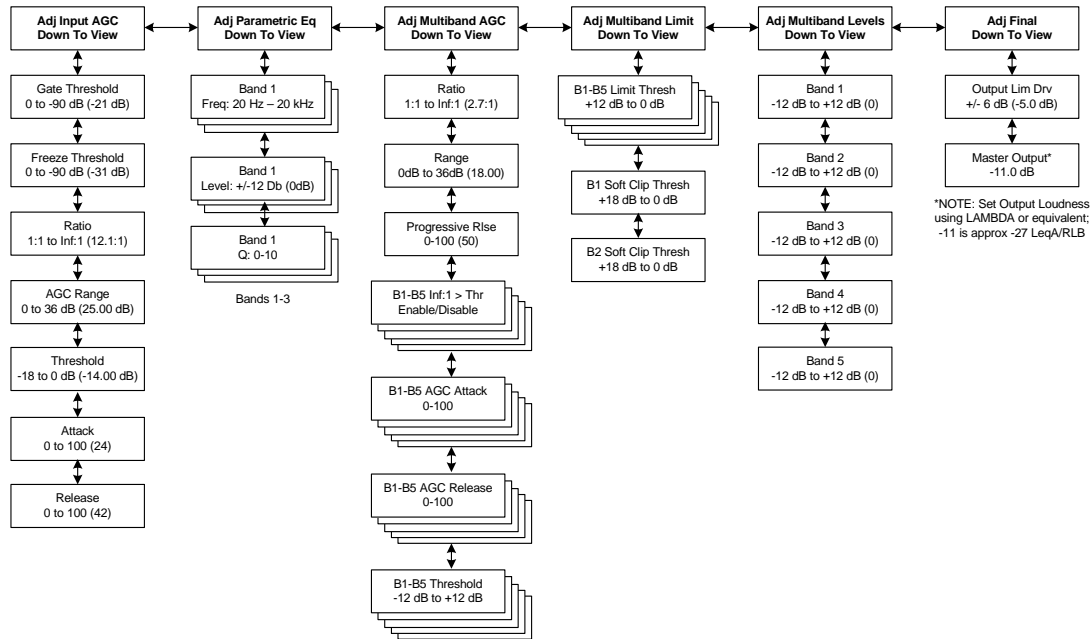


Figure 5-2 Adjust Processing Menus

But first... A Word on Our Crossovers

The multiband crossovers in the AERO.one consist of second order Linkwitz Reilly style filters that are 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

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 (default: -30dBFS)

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

- Freeze Thresh(old): 0dBFS to -90dBFS (default: -42dBFS)

-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: *Adjust Gate and Freeze 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 (default: 12.0:1)

- Range: 0dB - 36 dB (default: 24dB)

-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 (default: -16dBFS)
- Attack: 0 - 150, slowest - fastest (default: 21)
- Release: 0 - 150, slowest - fastest (default: 47)
- Progressive Release: 0 - 100, slowest - fastest (default: 50)

-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. Adjustable parameters are:

- Number of Bands: Five Bands/Four Bands (default: five bands)
- Ratio: 1.0:1 - Inf:1 (default: 3.0:1)
- B1-B5 Inf:1 Above Thresh: Enabled / Disabled (default: B1: Enabled, B2: Enabled, B3: Disabled, B4: Disabled, B5: 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 (default: 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 (default: 50)

-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 (defaults: B1:46, B2:88, B3:88, B4:88, B5:92)

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

- B1 - B5 AGC Release: 0 - 150, slowest - fastest (defaults: B1:50, B2:60, B3:86,

B4:88, B5:92)

-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 (default: 0.00dB all)

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

Multiband Limiters

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

- B1 - B5 Lim(it) Thresh(old): +12.00dB - 0.00dB (default: B1:+4.25dB, B2:+4.25dB, B3:+6.50dB, B4:+9.00dB, B5:+9.00dB)

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

- B1 Soft Clip Thresh(old): +12.00dB - 0.00dB (default: +3.00dB)

- B2 Soft Clip Thresh(old): +12.00dB - 0.00dB (default: +6.00dB)

-The point above which band one (low bass) is very quickly limited, acting more like a clipper without the artifacts. 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 (defaults: 0dB, all bands)

-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 (default: -5dB)

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

- Output Level: -36dB - 0dB (default: -11dB)

-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: *With TV 5B Gen selected and normal dialogue-based programming applied, loudness will measure approximately -24 LKFS using a Linear Acoustic LAMBDA or LQ-1000 or equivalent ITU-R BS.1770 loudness meter.*

| | TV_5B_Gen | TV_5B_Light | TV_5B_Heavy | TV_5B_Loud | ITU_Loud_Limit | Protection_Limit | SPORTS | MUSIC HVY |
|---------------------------|-----------|-------------|-------------|------------|----------------|------------------|-----------|-----------|
| Input AGC | | | | | | | | |
| Gate Threshold | -21 dB | -21 dB | -21 dB | -21 dB | -21 dB | -21 dB | -40 dB | -30 dB |
| Freeze Thresh | -31 dB | -31 dB | -31 dB | -31 dB | -42 dB | -31 dB | -55 dB | -42 dB |
| Ratio | 12.5:1 | 12.5:1 | 12.5:1 | 12.5:1 | 12.5:1 | 1.0:1 | 9.1:1 | 11.1:1 |
| AGC Range | 25.00 dB | 25.00 dB | 25.00 dB | 25.00 dB | 25.00 dB | 24.00 dB | 21.00 dB | 15.00 dB |
| Threshold | -14.00 dB | -14.00 dB | -14.00 dB | -14.00 dB | -14.00 dB | -16.00 dB | -12.00 dB | -15.00 dB |
| Attack | 24 | 24 | 24 | 24 | 10 | 24 | 21 | 20 |
| Release | 42 | 42 | 42 | 42 | 29 | 42 | 40 | 20 |
| Parametric EQ | | | | | | | | |
| B1 Frequency | 57 Hz | 57 Hz | 57 Hz | 57 Hz | 57 Hz | 57 Hz | 23 Hz | 57 Hz |
| B1 Gain | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 2.75 dB |
| B1 Q (Octaves) | 1.5 | 1.5 | 1.5 | 1.5 | 1.5 | 1.5 | 1.5 | 1.8 |
| B2 Frequency | 2.89 kHz | 2.89 kHz | 2.89 kHz | 2.89 kHz | 2.89 kHz | 2.89 kHz | 2.89 kHz | 2.51 kHz |
| B2 Gain | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 2.00 dB |
| B2 Q (Octaves) | 2.0 | 2.0 | 2.0 | 2.0 | 2.0 | 2.0 | 2.0 | 2.0 |
| B3 Frequency | 15.53 kHz | 15.53 kHz | 15.53 kHz | 15.53 kHz | 15.53 kHz | 15.53 kHz | 15.53 kHz | 12.59 kHz |
| B3 Gain | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 3.00 dB |
| B3 Q (Octaves) | 1.0 | 1.0 | 1.0 | 1.0 | 1.0 | 1.0 | 1.0 | 2.0 |
| Multiband AGC | | | | | | | | |
| Ratio | 2.8:1 | 1.4:1 | 5.9:1 | 14.3:1 | 1.0:1 | 1.0:1 | 2.4:1 | 12.5:1 |
| Range | 16.00 dB | 12.00 dB | 18.00 dB | 18.00 dB | 24.00 dB | 24.00 dB | 12.00 dB | 11.00 dB |
| Progressive Rel | 50 | 50 | 50 | 50 | 50 | 50 | 35 | 40 |
| Ratio Above Threshold: | | | | | | | | |
| B1 Inf. 1 | Enable | Enable | Enable | Enable | Disable | Disable | Enable | Disable |
| B2 Inf. 1 | Enable | Enable | Enable | Enable | Disable | Disable | Enable | Enable |
| B3 Inf. 1 | Enable | Enable | Enable | Enable | Disable | Disable | Enable | Enable |
| B4 Inf. 1 | Disable | Disable | Disable | Disable | Disable | Disable | Disable | Disable |
| B5 Inf. 1 | Disable | Disable | Disable | Disable | Disable | Disable | Disable | Disable |
| B1 AGC Attack | 22 | 22 | 22 | 22 | 46 | 46 | 22 | 36 |
| B2 AGC Attack | 32 | 32 | 32 | 32 | 88 | 88 | 43 | 36 |
| B3 AGC Attack | 42 | 42 | 42 | 42 | 88 | 88 | 43 | 40 |
| B4 AGC Attack | 46 | 50 | 56 | 62 | 88 | 88 | 55 | 38 |
| B5 AGC Attack | 66 | 66 | 75 | 75 | 92 | 92 | 60 | 32 |
| B1 AGC Release | 56 | 56 | 56 | 56 | 50 | 50 | 20 | 54 |
| B2 AGC Release | 67 | 67 | 67 | 67 | 60 | 60 | 32 | 58 |
| B3 AGC Release | 68 | 68 | 62 | 68 | 86 | 86 | 40 | 51 |
| B4 AGC Release | 74 | 74 | 67 | 74 | 88 | 88 | 38 | 54 |
| B5 AGC Release | 84 | 84 | 72 | 84 | 92 | 92 | 36 | 75 |
| B1 AGC Thresh | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 3.00 dB | 0.00 dB |
| B2 AGC Thresh | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 3.00 dB | 0.00 dB |
| B3 AGC Thresh | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 3.00 dB | 0.00 dB |
| B4 AGC Thresh | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 3.00 dB | 0.00 dB |
| B5 AGC Thresh | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 3.00 dB | 0.00 dB |
| Multiband Limiters | | | | | | | | |
| B1 Lim Thresh | 6.75 dB | 5.75 dB | 4.25 dB | 4.25 dB | 4.25 dB | 18.00 dB | 6.00 dB | 14.00 dB |
| B2 Lim Thresh | 9.00 dB | 8.00 dB | 4.25 dB | 4.25 dB | 7.00 dB | 18.00 dB | 6.00 dB | 10.00 dB |
| B3 Lim Thresh | 9.50 dB | 8.50 dB | 6.50 dB | 6.50 dB | 7.50 dB | 18.00 dB | 7.50 dB | 7.50 dB |
| B4 Lim Thresh | 11.25 dB | 10.25 dB | 9.00 dB | 9.00 dB | 9.25 dB | 18.00 dB | 7.25 dB | 8.50 dB |
| B5 Lim Thresh | 13.00 dB | 12.00 dB | 9.00 dB | 9.00 dB | 11.00 dB | 18.00 dB | 6.75 dB | 10.50 dB |
| B1 Soft Clip Thr | 6.00 dB | 6.00 dB | 6.00 dB | 6.00 dB | 6.00 dB | 18.00 dB | 3.00 dB | 6.00 dB |
| B2 Soft Clip Thr | 6.00 dB | 6.00 dB | 6.00 dB | 6.00 dB | 6.00 dB | 18.00 dB | 3.00 dB | 6.00 dB |
| Multiband Levels | | | | | | | | |
| B1 Out Mix | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | -3.00 dB | 1.25 dB |
| B2 Out Mix | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB |
| B3 Out Mix | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB |
| B4 Out Mix | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.25 dB |
| B5 Out Mix | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 0.00 dB | 1.00 dB |
| Final Stage | | | | | | | | |
| Output Lim Drv | -5.00 dB | -5.00 dB | -5.00 dB | -5.00 dB | 0.00 dB | -3.00 dB | -2.50 dB | -2.75 dB |
| Master Output | -11.0 dB | -11.0 dB | -11.0 dB | -11.0 dB | -6.0 dB | 0.0 dB | -11.0 dB | -10.0 dB |

Figure 5-3 Default Settings for Factory Presets

Chapter 6: Troubleshooting

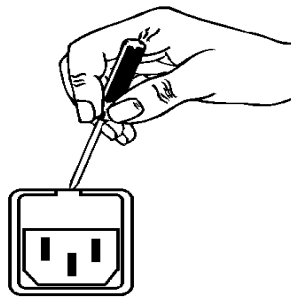
The Linear Acoustic AERO.one is a very stable and reliable unit, and most problems can be traced back to mis-wiring causing incorrect signals to be applied to the unit, or more than likely mis-configuration. As simple as the unit is to use once installed, it is very flexible and thus a necessarily complex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

6.1 Problems and Possible Causes

One of the best troubleshooting features of the AERO.one 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 I/O 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.

6.1.1 Unit won't power on

First check the obvious and make sure that the power cord or cords are plugged into live AC outlets (it happens). Next, see if either or both of the blue front panel PSU lights is illuminated. If not, check the power supply fuses by removing the power cords and using a small flat-blade screwdriver to remove the fuse holder from the IEC inlet module. The holder carries a spare fuse and extras were included in the packing kit with the unit. If the fuse is bad and continues to blow, please contact the factory.



6.1.2 The Ref LED is red

Check to make sure that an AES reference signal is connected to the unit and that the correct reference is selected. Note that this signal can be either AES black (i.e. an AES signal with silent audio), or an AES signal with audio. Reference can be derived from Main Input 1, Input 4, SDI, or Internal within the I/O Options menu (see Chapter 4).

6.1.3 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). If improper reference is chosen, the Ref LED will be red and the unit will revert to internal 48kHz reference allowing the 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.one is feeding is not properly referenced. Using the master by-pass function of the unit is a helpful way to troubleshoot this issue.

6.1.4 Received Audio Has Dropouts

The primary cause of this is the Dolby Digital AC-3 encoder not being properly referenced with respect to the MPEG video encoder/multiplexer. This is normally accomplished very simply by having all gear referenced to the local plant.

In some cases, such as with certain equipment from Tandberg, a special reference signal is generated in the multiplexer and it needs to be connected back to the Dolby Digital (AC-3) encoder and used as its reference. If the AERO.one contains the Dolby Digital encoder option, this can be accomplished by applying the signal to Ref In and selecting External Reference as the clock source (Setup -> I/O menu). If using an external Dolby Digital encoder such as the DP569, set it to "REF IN/SRC On mode. Possibly the best method is to re-embed the encoded Dolby Digital (AC-3) signal and apply it to the MPEG encoder via SDI. Consult your MPEG encoder reference manual for more details.

6.1.5 Audio Pumps and Breathes

Sometimes audible in less sophisticated processors or when attack and release times are not adjusted properly. The factory presets included with the AERO.one will not cause pumping or breathing, but if it sounds like this is happening, there are three common possible causes. First, station audio could be inadvertently pre-processed by legacy gear. The AERO.one performs best when it receives unprocessed audio. Next, the AERO.one could be inadvertently upmixing a program that is actually 5.1 channels (this will also cause loss or damage to dialogue). Finally, if you are using an external Dolby Digital encoder, make sure that the Audio Coding Mode (acmod) is set to 3/2L or 5.1 channels. If inadvertently left in 2/0 or stereo mode, only the Left front and Right front channels would be encoded.

NOTE: *If the AERO.one is installed using an external Dolby Digital (AC-3) encoder make absolutely certain that dialnorm has been set correctly and that a setting called "RF Overmod Protect" is set to Disabled or Off. Do not confuse this with RF Mode DRC which should be set to Film Standard.*

6.1.6 Nielsen Watermark Issues

Careful testing has shown that the Nielsen audience measurement watermark is not disturbed by any of the factory presets, nor upmixing when the watermarking equipment is installed prior to the AERO.one. In fact, improvements in code recovery have been noted when Linear Acoustic audio processing is installed after watermark insertion. Therefore, we now recommend that watermarking gear be installed ahead of any audio processing. This is a change from previous recommendations that suggested installation between processing and Dolby Digital (AC-3) encoding. Installing prior to upmixing allows over-writing of the watermark to be performed more effectively than when watermarking is inserted after upmixing.

6.1.7 Problems with EAS decoding

It is important to locate EAS encoding equipment after any audio processing, including AERO.one. It is critical that processing be minimized for use with current EAS systems. The AERO.one provides a special “post processing” insert specifically for EAS audio and it should be used, especially if the Dolby Digital encoder option is installed.

Chapter 7: Specifications

Table 7-1 Electrical Specifications

| | |
|------------------------------|---|
| Sampling Rate | 48 kHz ($\pm 0.1\%$) |
| Processing Delay | 15 msec fixed, Main Out 1-6 16 msec fixed, LoRo Out 7/8 21 msec fixed, LtRt Out 7/8 Dolby Digital/DD Plus encoding adds 137msec to the total. |
| Processing Algorithms | Linear Acoustic AEROMAX loudness and DRC engine plus UPMAX upmixing engine; ITU downmix |
| Audio Word Length | 24-bit performance |
| Digital Audio In | Four BNC female connectors, AES-3ID-2001/SMPTE 276M |
| Digital Audio Out | Four BNC female connectors, AES-3ID-2001/SMPTE 276M |
| 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 |
| Digital I/O | |
| Digital Audio Inputs | Six unbalanced female BNC connectors, comply with AES-3ID-2001/SMPTE 276M. Internal 75-Ohm termination. |
| Digital Audio Outputs | Six unbalanced female BNC connectors that comply with AES-3ID-2001/SMPTE 276M specifications. |
| SDI I/O | |
| HD/SD-SDI I/O | Up to 16 channels of audio can be de-muxed from any of the four groups and 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. |

Table 7-2 Audio Encoding Specifications

| | |
|----------------------------|---|
| Audio Encoder Types | Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) per ATSC A/52:2010 |
| Encoder Modes | Primary: up to 5.1; Secondary: up to 2.0 |
| Encoder Rates | Dolby Digital: Up to 640 kbps Dolby Digital Plus: Up to 1532 kbps |
| Encoder Latency | 137 msec fixed |

Table 7-3 Mechanical Specifications

| | |
|---------------------------|---|
| Dimensions | 1.75 × 19 × 24 in (44.4 × 48.3 × 61 cm) |
| Net Weight | 6 lbs (2.72 kg) approx. |
| Shipping Weight | 8 lbs (3.63 kg) approx. |
| Power Requirements | 90-240VAC 50–60 Hz |
| Power Consumption | 40 W maximum |

Table 7-4 Environmental Specifications

| | |
|--|---|
| Operating Temperature | 0°C to 50°C |
| 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 7-5 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 10 or 100BASE-T.

GPI/O Parallel Control Port

TTL level controls, active Low. The 5-V output for external GPO indicators is limited by a self-resetting fuse.

Table 7-6 GPI/O Port Pinout

| Pin | Connection |
|-----|---------------|
| 1 | GPI 1 |
| 2 | GPI 2 |
| 3 | GPI 3 |
| 4 | GPI 4 |
| 5 | Ground/Common |
| 6 | GPO 1 |
| 7 | GPO 2 |
| 8 | GPO 3 |
| 9 | GPO 4 |