LINEAR ACOUSTIC

AERO.air

(5.1) and (DTV) Versions

Digital Television Audio Processor with HD/SD-SDI, Dolby Digital (AC-3) Encoding and Dolby E/Dolby Digital (AC-3) Decoding and Nielsen Watermark Encoding

User Guide

L I N E A R   A C O U S T I C

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User Guide

Release Date: March, 2011
Software Version: 1.18.01 2U and later

Warranty Information

DISCLAIMER OF WARRANTIES: Products manufactured by Linear Acoustic are warranted against defects in material and workmanship under the standard Telos Alliance 5-year warranty from the date of purchase. **THERE ARE NO OTHER IMPLIED OR EXPRESS WARRANTIES AND NO WAR- RANTY 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 ACCI- DENTS, UNAUTHORIZED CUSTOMER REPAIRS, OR ANY FURTHER DAMAGE CAUSED BY INADEQUATE PACKAGING FOR SERVICE RETURN ARE NOT COV- ERED 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 processing and is manufactured under license.

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and/or decoding and is manufactured under license from Dolby Laboratories, Inc. The encoder contains confidential unpublished works and is protected by US and foreign patents and is copyright 2007 Dolby Laboratories, Inc.

This product optionally contains Nielsen Watermarking technology and is manufactured under license.

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Regulatory Notices and Fusing Information

FCC

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with this instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his or her own expense.

Canada

This Class A digital apparatus complies with Canadian ICES-003.

UL

WARNING: Troubleshooting must be performed by a trained technician. Do not attempt to service this equipment unless you are qualified to do so.

Check that the correct fuses have been installed. To reduce the risk of fire, replace only with fuses of the same type and rating.

Exposed portions of the power supply assembly are electrically "hot". In order to reduce the risk of electrical shock, the power cord MUST be disconnected when the power supply assembly is removed.

The ground terminal of the power plug is connected directly to the chassis of the unit. For continued protection against electric shock, a correctly wired and grounded (earthed) three-pin power outlet must be used. Do not use a ground-lifting adapter and never cut the ground pin on the three-prong plug.
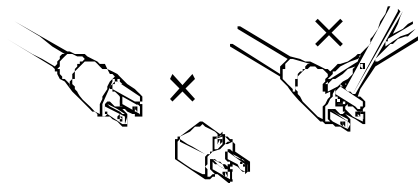
UK

As the colours of the cores in the mains lead may not correspond with the coloured markings identifying the terminals in your plug, proceed as follows:

- The core that is coloured green and yellow must be connected to the terminal in the plug identified by the letter E or by the earth symbol ⏚ or coloured green or green and yellow.
- The core that is coloured blue must be connected to the terminal that is marked with the letter N or coloured black.
- The core that is coloured brown must be connected to the terminal that is marked with the letter L or coloured red.
- This apparatus must be earthed.

EU

This equipment complies with the EMC requirements of EN55103-1 and EN55103-2 when operated in an E2 environment in accordance with this manual.

IMPORTANT SAFETY NOTICE

This unit complies with the safety standard EN60065. The unit shall not be exposed to dripping or splashing and no objects filled with liquids, such as coffee cups, shall be placed on the equipment. To ensure safe operation and to guard against potential shock hazard or risk of fire, the following **must** be observed:

o   Ensure that your mains supply is in the correct range for the input power requirement of the unit.
o   Ensure **fuses** fitted are the **correct rating and type** as marked on the unit.
o   The unit **must be earthed** by connecting to a correctly wired and **earthed** power outlet.
o   The **power cord** supplied with this unit must be wired as follows:

        Live—Brown   Neutral—Blue   Earth—Green/Yellow

(GB)

IMPORTANT – NOTE DE SECURITE

Ce materiel est conforme à la norme EN60065. Ne pas exposer cet appareil aux éclaboussures ou aux gouttes de liquide. Ne pas poser d'objets remplis de liquide, tels que des tasses de café, sur l'appareil. Pour vous assurer d'un fonctionnement sans danger et de prévenir tout choc électrique ou tout risque d'incendie, veillez à observer les recommandations suivantes.

o   Le selecteur de tension doit être placé sur la valeur correspondante à votre alimentation réseau.
o   Les fusibles doivent correspondre à la valeur indiquée sur le materiel.
o   Le materiel doit être correctement relié à la terre.
o   Le cordon secteur livré avec le materiel doit être câblé de la manière suivante:

        Phase—Brun   Neutre—Bleu   Terre—Vert/Jaune

(F)

WICHTIGER SICHERHEITSHINWEIS

Dieses Gerät entspricht der Sicherheitsnorm EN60065. Das Gerät darf nicht mit Flüssigkeiten (Spritzwasser usw.) in Berührung kommen; stellen Sie keine Gefäße, z.B. Kaffeetassen, auf das Gerät. Für das sichere Funktionieren des Gerätes und zur Unfallverhütung (elektrischer Schlag, Feuer) sind die folgenden Regeln unbedingt einzuhalten:

o   Der Spannungswähler muß auf Ihre Netzspannung eingestellt sein.
o   Die Sicherungen müssen in Typ und Stromwert mit den Angaben auf dem Gerät übereinstimmen.
o   Die Erdung des Gerätes muß über eine geerdete Steckdose gewährleistet sein.
o   Das mitgelieferte Netzkabel muß wie folgt verdrahtet werden:

        Phase—braun   Nulleiter—blau   Erde—grün/gelb

(D)

NORME DI SICUREZZA – IMPORTANTE

Questa apparecchiatura è stata costruita in accordo alle norme di sicurezza EN60065. Il prodotto non deve essere sottoposto a schizzi, spruzzi e gocciolamenti, e nessun tipo di oggetto riempito con liquidi, come ad esempio tazze di caffè, deve essere appoggiato sul dispositivo. Per una perfetta sicurezza ed al fine di evitare eventuali rischi di scossa êlettrica o d'incendio vanno osservate le seguenti misure di sicurezza:

o   Assicurarsi che il selettore di cambio tensione sia posizionato sul valore corretto.
o   Assicurarsi che la portata ed il tipo di fusibili siano quelli prescritti dalla casa costruttrice.
o   L'apparecchiatura deve avere un collegamento di messa a terra ben eseguito; anche la connessione rete deve avere un collegamento a terra.
o   Il cavo di alimentazione a corredo dell'apparecchiatura deve essere collegato come segue:

        Filo tensione—Marrone   Neutro—Blu   Massa—Verde/Giallo

(I)

AVISO IMPORTANTE DE SEGURIDAD

Esta unidad cumple con la norma de seguridad EN60065. La unidad no debe ser expuesta a goteos o salpicaduras y no deben colocarse sobre el equipo recipientes con liquidos, como tazas de cafe. Para asegurarse un funcionamiento seguro y prevenir cualquier posible peligro de descarga o riesgo de incendio, se han de observar las siguientes precauciones:

o   Asegúrese que el selector de tensión esté ajustado a la tensión correcta para su alimentación.
o   Asegúrese que los fusibles colocados son del tipo y valor correctos, tal como se marca en la unidad.
o   La unidad debe ser puesta a tierra, conectándola a un conector de red correctamente cableado y puesto a tierra.
o   El cable de red suministrado con esta unidad, debe ser cableado como sigue:

        Vivo—Marrón   Neutro—Azul   Tierra—Verde/Amarillo

(E)

VIKTIGA SÄKERHETSÅTGÄRDER!

Denna enhet uppfyller säkerhetsstandard EN60065. Enheten får ej utsättas för yttre åverkan samt föremål innehållande vätska, såsom kaffemuggar, får ej placeras på utrustningen." För att garantera säkerheten och gardera mot eventuell elchock eller brandrisk, måste följande observeras:

o   Kontrollera att spänningsväljaren är inställd på korrekt nätspänning.
o   Konrollera att säkringarna är av rätt typ och för rätt strömstyrka så som anvisningarna på enheten föreskriver.
o   Enheten måste vara jordad genom anslutning till ett korrekt kopplat och jordat el-uttag.
o   El-sladden som medföljer denna enhet måste kopplas enligt följande:

        Fas—Brun   Neutral—Blå   Jord—Grön/Gul

(S)

BELANGRIJK VEILIGHEIDS-VOORSCHRIFT:

Deze unit voldoet aan de EN60065 veiligheids-standaards. Dit apparaat mag niet worden blootgesteld aan vocht. Vanwege het risico dat er druppels in het apparaat vallen, dient u er geen vloeistoffen in bekers op te plaatsen. Voor een veilig gebruik en om het gevaar van electrische schokken en het risico van brand te vermijden, dienen de volgende regels in acht te worden genomen:

o   Controleer of de spanningscaroussel op het juiste Voltage staat.
o   Gebruik alleen zekeringen van de aangegeven typen en waarden.
o   Aansluiting van de unit alleen aan een geaarde wandcontactdoos.
o   De netkabel die met de unit wordt geleverd, moet als volgt worden aangesloten:

        Fase—Bruin   Nul—Blauw   Aarde—Groen/Geel

(NL)

Fusing Information

⚠️ *WARNING:* **To reduce the risk of fire, replace fuses only with the same type and rating.**

Both units use a universal switching power supply that handles the full range of nominal mains voltages between 90 and 264 VAC and any frequency between 50 Hz and 60 Hz.

Check Main Fuse

The Main fuse rating is:

T 1A L (1 Amp, 250 V, 20 mm, time-lag, low breaking capacity) for all operating voltages.

⚠️ *WARNING:* **The power to the unit must be off when the following steps are performed. Ensure that the main power cable to the unit is not connected to a power source.**

1. Open the fuse compartment door in the AC power input housing with a small flat-blade screwdriver (Figure 1). Carefully pull out the fuse carrier.
2. Check that the replacement fuse has the correct rating. The fuse carrier must be inserted into the compartment carefully. *Do not force the carrier into the compartment or both could be damaged.*
3. Snap the fuse compartment door closed.
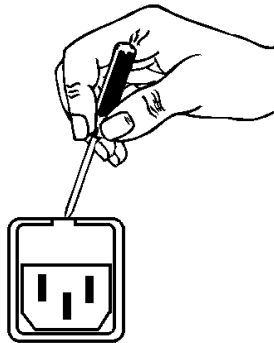
Figure 1 Checking the Main Fuse

Internal Fuse

The switching power supply contains a separate fuse. Most fault conditions should be protected by the main fuse.

If you find it necessary to replace the internal fuse, be certain to replace it with a fuse of the same type and rating as printed on the switching power supply board.

Chapter 1: Introduction

The Linear Acoustic AERO.air Audio/Loudness Manager is a state-of-the-art adaptive multiband broadcast audio/loudness processor. Available in two versions, the AERO.air (5.1) is capable of accepting up to 10 channels of audio while the AERO.air (DTV) handles up to 6 channels of audio. Both versions can apply loudness control and upmixing. The unit outputs 5.1 processed channels, as well as a full time LtRt version of the main audio channels, and the AERO.air (5.1) version can support up to two additional separate stereo or mono programs.

AERO.air includes metadata functionality where professional audio metadata (i.e. Dolby Metadata) can be applied to the unit to minimize processing and control functions such as upmixing automatically. AERO.air protects against loss of incoming metadata with reversion modes that allow application of plant reference loudness values. The resulting audio is clean, consistent, and appropriately dynamic - perfect for transmission to all consumers via Dolby Digital (AC-3) or any other format.

The AERO.air includes the following features:

- AEROMAX® multiband loudness processing provides ITU compliance for up to ten channels of audio; includes input AGC, parametric EQ, multiband AGC and limiting, look-ahead peak limiting, and selectable multiband source noise reduction.

- ITU-R BS.1770 meter for each output program simplifies loudness target calibration

- Support for 5.1+2+2, 5.1+2+Local, or 2+2+2+2 program configurations for AERO.air (5.1) and 5.1+Downmix for AERO.air (DTV).

- Selectable and adjustable upmixing of Main Input 1/2 (Left/Right front) via industry standard UPMAX® and the new UPMAX II two-channel to 5.1 channel upconversion algorithm for even more spacious results.

- EAS/AUX input and LtRt or LoRo downmix output available full-time

- HD/SD-SDI I/O for embedded audio and VANC metadata

- CrowdControl dialogue protection processing for 7/8 output

- Dual, redundant, auto ranging power supplies;

- Optional internal Dolby Digital (AC-3)/Dolby Digital Plus/Dolby E decoding and dual 5.1+2 Dolby Digital (AC-3)/Dolby Digital Plus encoding

- Optional Nielsen® Watermark Encoding

1.1   Principles of Operation

The AERO.air (5.1) processing structures support up to ten channels of main audio input. Two channel audio can be applied to the first Main Input (1/2) and it can be upmixed to 5.1 channels if desired. Local audio can be applied to Main Input 7/8, processed and up-mixed if desired, then used to voice-over or replace main program audio. The AERO.air (DTV) supports up to 6 channels of main audio input and produces 5.1 channels plus a stereo downmix output.

Specific processing presets and adjustments are discussed in Chapter 7.

Figure 1-1 shows the internal audio path of the AERO.air (5.1). 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.
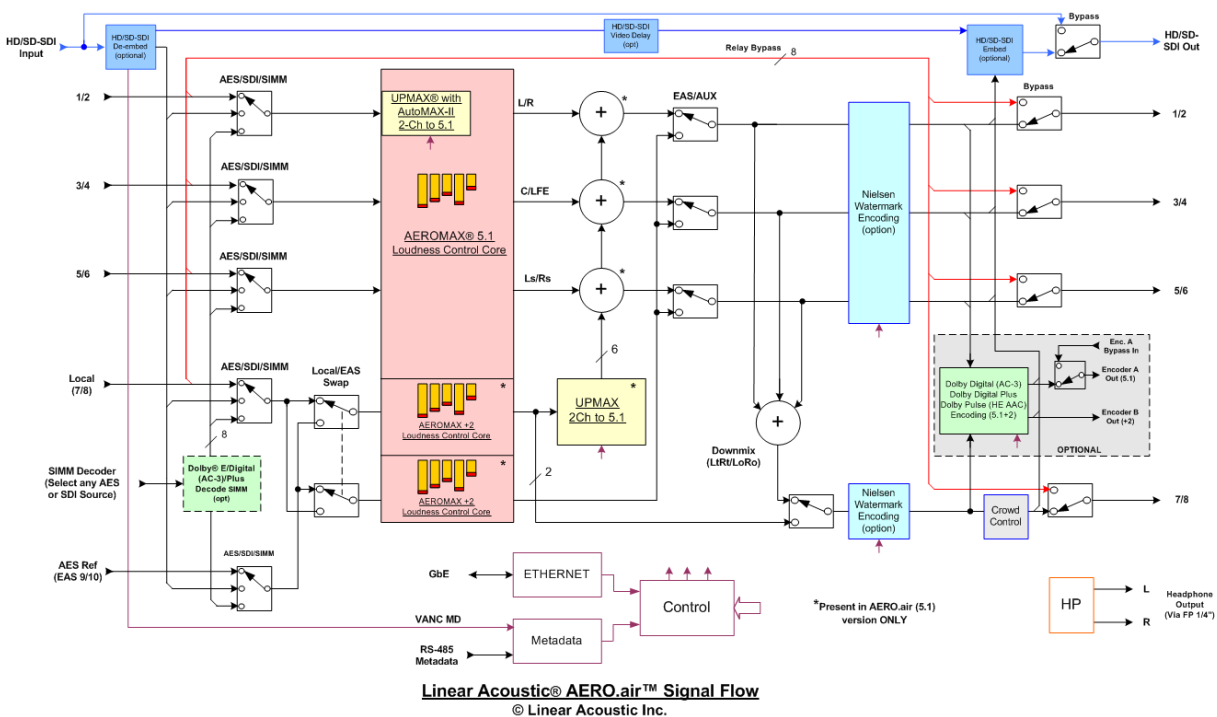
Figure 1-1 AERO.air (5.1) Audio Block Diagram

1.2   Location of AERO.air

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.air near the final emission DTV encoder/multiplexer as the units work closely together and will benefit from short cable runs and common clocking.

1.3   Reference Levels and Line Up Features

The AERO.air 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. The AERO.air contains a host of built in signal generation capabilities that are useful for both level calibration of downstream equipment, and for channel identification. Please see Chapter 4 for detailed information.

1.4   Software Updates

The USB connector located on the rear panel of the AERO.air is used for software updates and feature upgrades. Linear Acoustic will provide the proper USB memory device, software, and upgrade instructions as needed. Please do not insert any device into this connector without first contacting the factory as it may impair the performance of the unit and cause loss of on-air audio.

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

1.5   Powering Up and Shutting Down

After first connecting power to the dual power supplies, simultaneously and *briefly* depress the Left, Up, and Next buttons to start the unit. The front panel Linear Acoustic logo will illuminate blue and the unit will power up, taking about ninety seconds. To power down, simultaneously and briefly depress the Left, Up, and Next buttons and the unit will power down. This may take about 15 seconds so that all settings can be properly stored. Once the logo is no longer illuminated, the unit is completely powered down.

1.6   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.

-NOTES-

Chapter 2: Connections and Quick Setup

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

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:

    - Two IEC power cords (style matches country of order);

    - one DB-25 male connector and backshell for GPI/O connection;

    - one USB drive for future software updates (please save!);

    - 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.air installation requires:

- three standard rack space units with ADEQUATE VENTILATION (the unit vents from the front right side intake slots and the rear option panel and out through the left front exhaust fan;

- 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.air contains its electrical I/O.

2.3.1   Connection Ports

All of the AERO.air's connections are on the rear panel and are described in detail below. See Chapter 5: *Specifications* for specific pinouts.
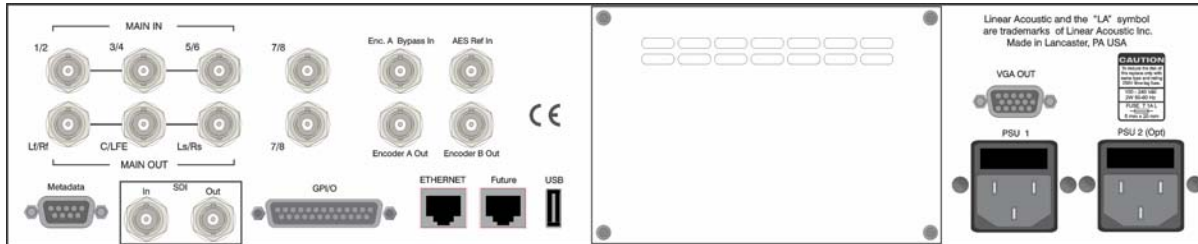
Figure 2-1 Rear Panel

- **Main 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, and 7/8 = Stereo +2 or Local input.

- **Encoder A Bypass In:** Connect the output of an optional external Dolby Digital (AC-3) encoder to this input for failover operation. If the unit enters bypass, the externally encoded signal will be relay bypassed through to the Encoder A Output.

- **AES REF (9/10 EAS Input):** 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.air. This connector is also used for the 9/10 Input.

NOTE: Appropriate reference must be selected for proper operation.

- **Main Audio Output:** Main 48kHz processed digital audio outputs with same channel configuration as the inputs on 1-6. Channels 7/8 contain a compatible LtRt downmix of the main program audio to which CrowdControl can be applied. In the AERO.air (5.1) version, 7/8 can be switched to the output of a stereo +2 program.

- **Encoder A Output:** Outputs up to 5.1 channels of audio encoded as a Dolby Digital (AC-3) signal over AES. The source for this encoder is the Main Out 1-6, so any program audio on these outputs will be Dolby Digital (AC-3) encoded). When in bypass, this connector is passively connected to the Bypass Input BNC directly above it.

- **Encoder B/9/10 Output:** Outputs mono or stereo audio encoded as Dolby Digital (AC-3) and the source for this encoder is Main Out 7/8. In the I/O menu of the AERO.air (5.1) version, this connector can be set to 9/10 output for a second +2 program or a downmix of the main.

- **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 10 PCM outputs (8 in the case of AERO.air (DTV) and both AC-3 encoder outputs can be embedded into an applied HD/SD-SDI signal. 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:** Connect to the GPI/O outputs of your routing or master control switcher to control special processor functions. Note that GPI A and B are used to control functions such as Upmixing and EAS processing bypass and require a held contact closure to ground for the duration of the function.

- **ETHERNET:** Automatically senses 100 or 1000BASE-T Ethernet connectivity. This connector is not currently supported an as such it is recommended that it not be connected to any network.

- **Future:** Extra RJ-45 connector for future I/O use.

- **PSU 1 and PSU 2:** Apply 100-240 VAC filtered power to BOTH power supplies. Essentially, two power supplies are used to "sum" AC power from two different sources thus improving plant reliability. Only one supply is required for continuos operation, but internally, the supplies operate in parallel and share the load which lowers heat.

2.4   Quick Setup Notes

The Linear Acoustic AERO.air 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 Dolby Digital (AC-3) encoded versions of the main and auxiliary programs.

The best way to learn the processor 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 depending on program content. Modern broadcast television offers the ability to process less, and even the heaviest presets in the AERO.air are moderate by comparison to what might be found in traditional NTSC audio processors.

2.5   Password Programming

After setup, it is probably a good idea to password-protect the AERO.air. This will prevent unauthorized or accidental changes to the settings. Please do not forget the password!

2.6   Video Delay Option

Audio processing, encoding and decoding and audience measurement watermark insertion each have latencies that when combined can be significant. While many installations can simply back off the audio delay added by all MPEG encoders (to make up for video encode latency), not all facilities are co-located with their transmission system. In many cases, transmission is handled by a completely separate company.

The video delay option provides a fixed 250 msec delay to the video and ciliary data portions of an applied HD or SD-SDI signal while the embedded audio channels receive no delay. In this manner, latency from processing, coding, and watermarking is taken into account. Timing the audio path to exactly 250msec can be accomplished in the I/O menu where utility delays can be set upon commissioning.

Chapter 3: Applications

The AERO.air 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 plus one showing how the unit is commonly used at cable and satellite facilities:

- Dolby Digital (AC-3) (PBS, FOX)

- Baseband PCM (NBC/ABC via MPEG1 Layer 2)

- PCM/Dolby E within HD-SDI (CBS/Univision)

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.air can be integrated within each of the environments.

> **NOTE:** The AERO.air (DTV) accepts up to six channels to handle 5.1 or stereo, and outputs 5.1 plus a downmix. The drawings below show the fully-featured AERO.air (5.1) which can handle up to 10 input channels.

AC-3 Distribution System

Figure 3-1 shows how an AERO.air would be installed in a typical AC-3 distribution system where the network is using AC-3 to deliver the 5.1-channel audio via a standard 19.39 Mbps transport stream. In the case of FOX network, 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.

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.air so that it matches these target metadata defaults. See section 3.3 below for a description of exactly how to set these parameters.
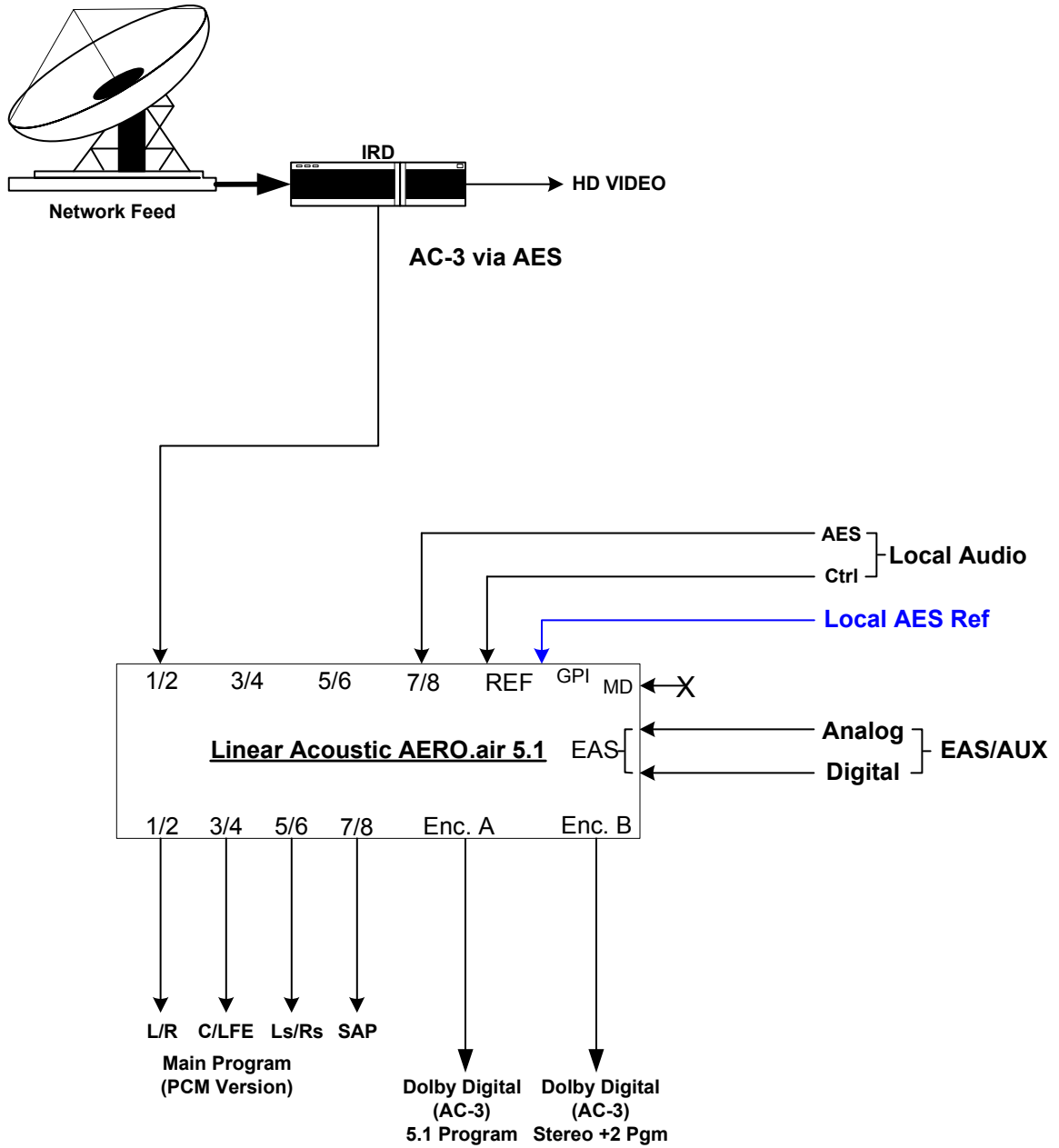
Figure 3-1 AERO.air (5.1) at a station receiving baseband AC-3

The transport stream approach is shown in Figure 3-2:
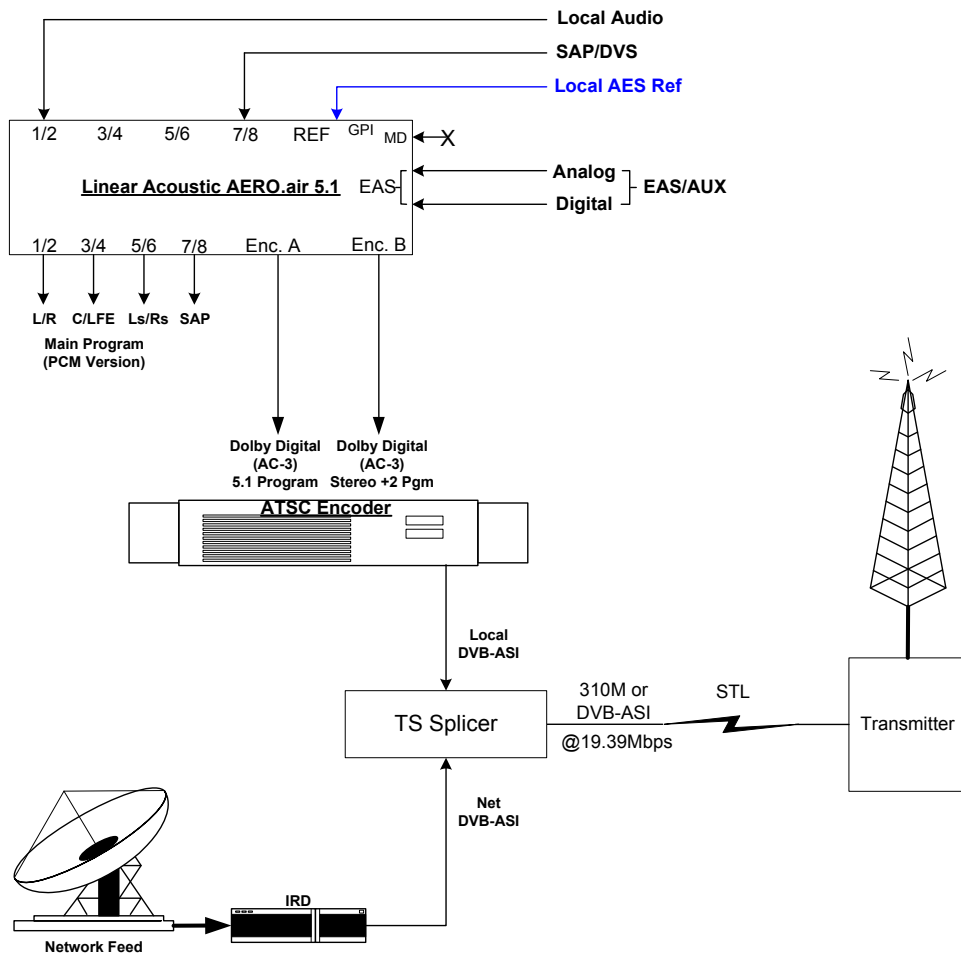
Figure 3-2 AERO.air (5.1) with Transport Stream Splicer

Note that because the network signal contains pre-encoded AC-3 audio, in this scenario no corrections or modifications can be made. The AC-3 signal could be extracted from the network signal via the IRD and processed if deemed necessary.

PCM Distribution System

Figure 3-3 shows how an AERO.air 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 ABC and NBC, MPEG 1 Layer 2 audio coding 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. Metadata is not presently used by either network. Exact channel configuration may differ between networks.
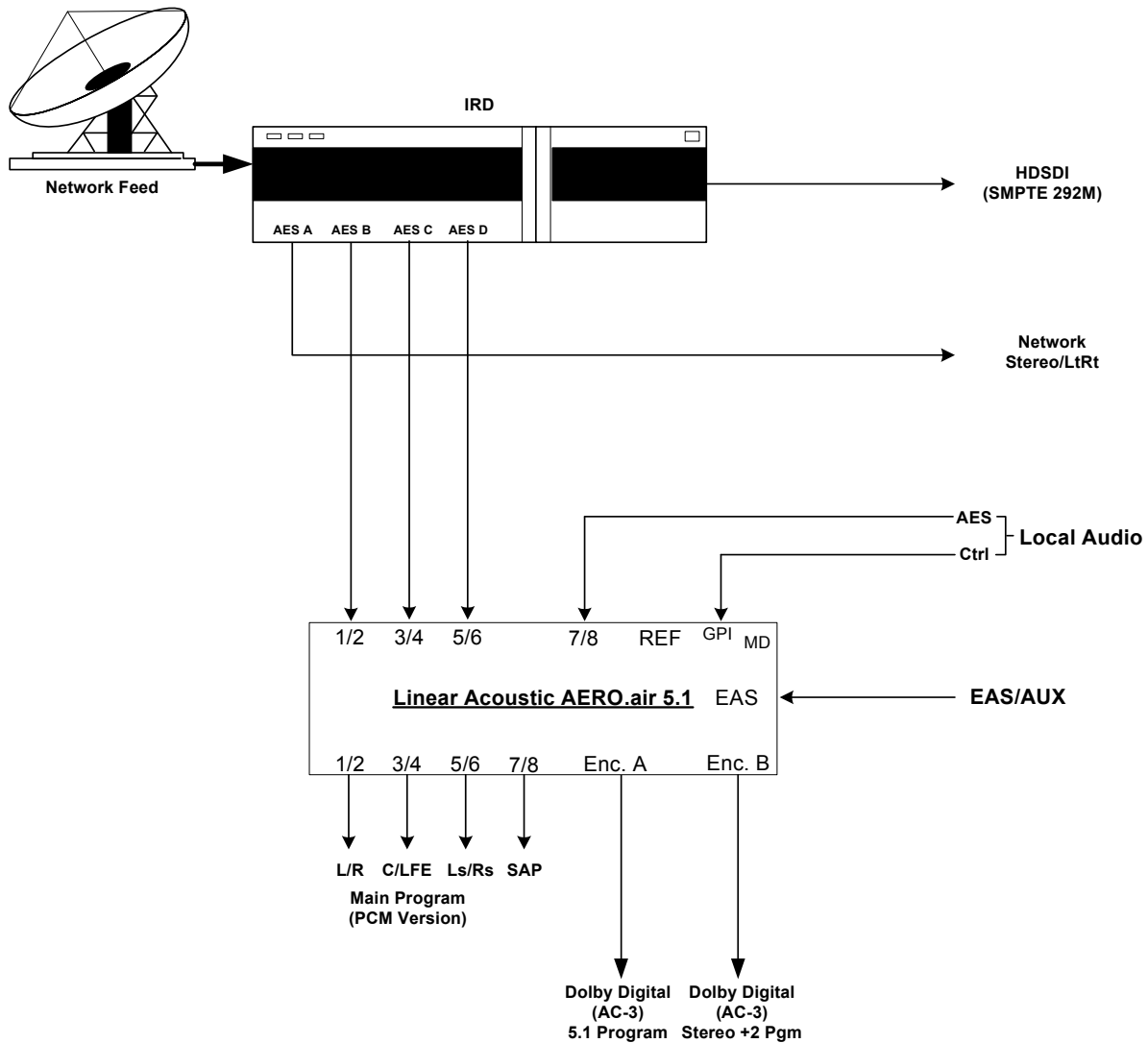
Figure 3-3 AERO.air (5.1) in a typical baseband PCM system (i.e. ABC and NBC).

Note that as shown above, network and local audio can get completely independent audio processing and only what is necessary for each. GPI can smoothly control this.

HD-SDI Distribution System

Figure 3-4 shows how an AERO.air would be installed in a typical HD-SDI environment, similar to that in use by CBS network. In this case, professional audio metadata is de-embedded from the HD-SDI signal and is passed to the AERO.air where it can be used to control upmixing.
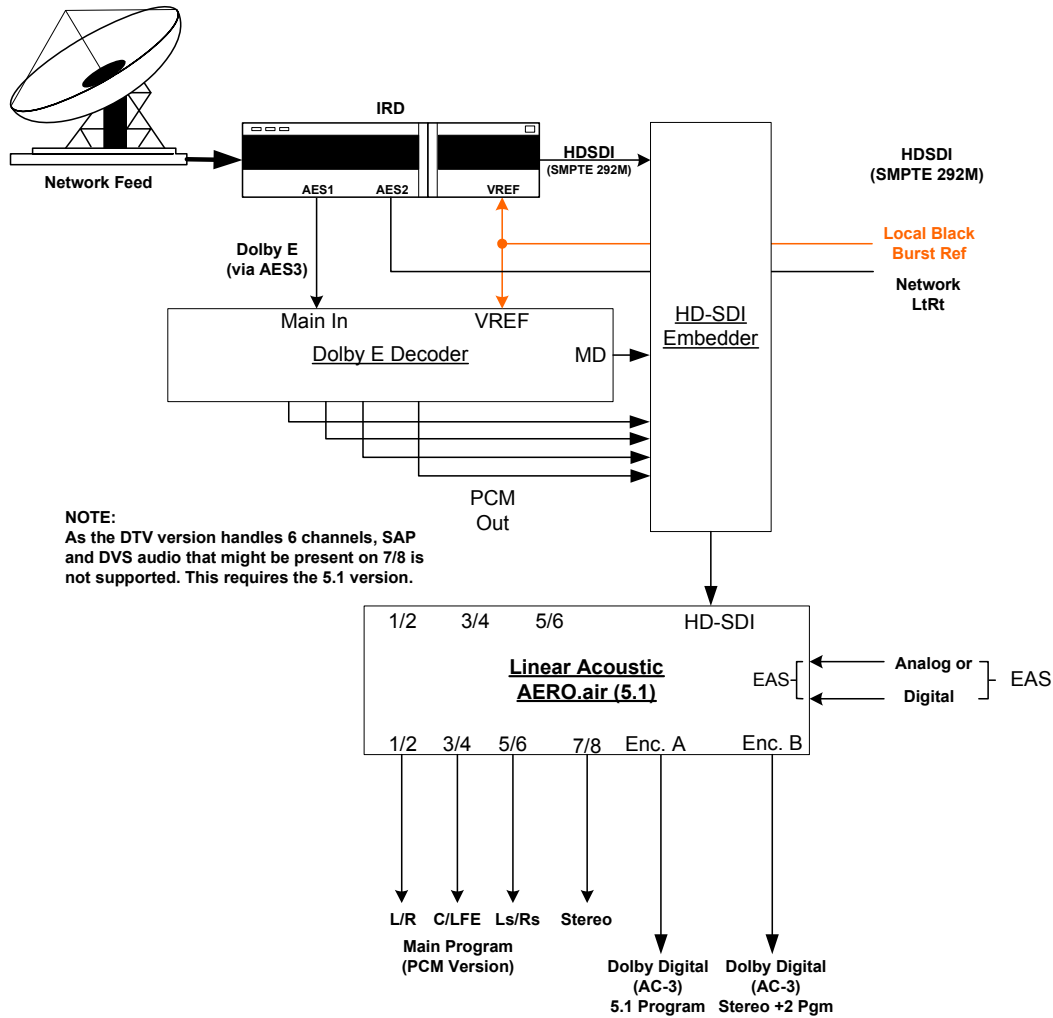
Figure 3-4 AERO.air in a typical HD-SDI environment (CBS and others)

The Dolby Digital encoder is set to default values for audio coding mode, dialogue level, and Line and RF Mode DRC presets as described below. CBS network uses Dolby E to carry the audio portion of their HD network feed, and embeds the professional audio metadata along with closed captioning and other data in the VANC space of an HD-SDI stream. After receiving the network signal, local stations also have the choice of keeping the signals in baseband format. Note that the REF signal for the AERO.air can be obtained easily from the unused PCM Delay output of the DP572, thereby simply locking it to the plant video reference.

All three scenarios show how simply the AERO.air integrates into the facility. All three systems also show the product's important "Local" features. Stereo or mono local audio can be applied to the AERO.air via either two analog inputs (balanced XLR-type connectors) or a 75-Ohm AES3-ID input (unbalanced BNC). A stereo and matrix surround-compatible LtRt signal is also generated from the main audio program and provided as both a balanced analog and an unbalanced AES3-ID output.

3.1   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.air, audio metadata can be used to control functions such as when to apply upmixing and eventually guide processing preset selection. Use of an AERO.air with metadata can also protect stations from metadata failure.

3.2   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.air can work in tandem with this data to ensure audio of the highest reliability and consistency. When installing an AERO.air, network audio metadata should be connected only to the AERO.air.

By analyzing the incoming metadata, features such as upmixing can be automatically invoked when appropriate. The AERO.air 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.air, 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.air will automatically pre-adjust the audio to match these settings.

WARNING: For external Dolby Digital encoders, make certain that "RF Overmod Protect" is set to Off or Disabled. Although its name makes it sound like a useful feature, it is not and WILL produce downmixed audio that pumps.

3.3   Dolby Digital Encoder Setup (local metadata)

To use the AERO.air 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 a dialogue loudness meter such as a Linear Acoustic LAMBDA or LQ-1000 or equivalent ITU loudness meter.

- **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.

3.3.1   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.air can be relied upon to keep all programming in 5.1 channel surround sound.

3.3.2   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.air 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.air is doing the job of matching each program to a single loudness value.

To set Dialogue Level use of a Linear Acoustic LAMBDA or LQ-1000 makes it 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.

3.3.3   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.air preset such as TV 5B Gen 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. The AERO.air will still react to badly mismatched loudness levels, but will do so in combination with the DRC profile in the Dolby Digital encoder.

NOTE: *Do not pick "None" mode because contrary to its name, it is not!*

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.air in TV 5B Gen)

- Channel Mode (acmod): 3/2L

- Line Mode and 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.

3.3.4   Audience Measurement Systems

Audience measurement systems such as the ones available from Nielsen 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, external 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.

However, the best answer is the Nielsen Watermark option which enables the AERO.air to insert the Nielsen Watermark(s) internally at exactly the right place in the signal path: after processing and prior to Dolby Digital (AC-3) encoding. The internal Nielsen Watermark option is especially useful if the Dolby decoding option is also installed as input and output audio are encoded and expensive and potentially unreliable external equipment is eliminated.

3.3.5   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.air unit is configured. Main audio is muted and externally supplied EAS programming is inserted into the Left and Right output channels post processing.

Chapter 4: Detailed Operation

This chapter discusses in detail the structure of the Linear Acoustic AERO.air, how to use the front panel LCD 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 5.

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 top level of the main screen looks like the right side of Figure 4-1. The meters show input, processing, and output activity for the Main Inputs. The submenus to the right of the meters are used to access the processor's functions, which are described in this chapter.
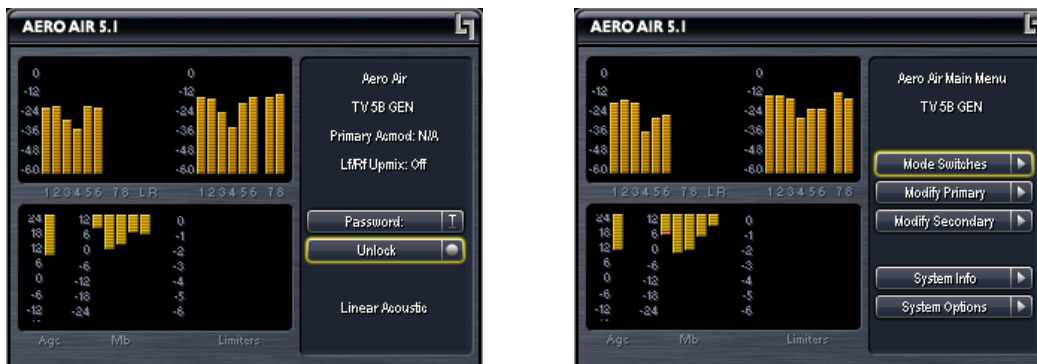
Figure 4-1 Locked Main Screen (left); Unlocked Main Screen (right)

4.2   Mode Switches

Mode switches adjust parameters for features such as upmixing, bass enhancement, and the CrowdControl option.

- **LfRf Upmix (On/Auto/Off) -** When the unit is not under GPI control (System Options -> GPI Setup), this is a control allowing manual invoking of standard upmixing and auto-upmixing. When the unit is under GPI control, this is an indication of the current mode of the upmixer: On, Auto, or Off.

4.2.1   Upmixing

Upmixing is the process whereby two channel audio can be convincingly re-mapped into 5.1 channels using sophisticated level and phase detection techniques. The proprietary, industry standard and road-proven UPMAX$^®$ algorithm is incorporated in the AERO.air, and optionally the new UPMAX$^®$ II algorithm can be added for enhanced spaciousness and resilience to low bitrate source material. The UPMAX family enables 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) from any encoding system. 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

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. UPMAX II excels at upmixing music of all genres.

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.

- **Upmix LfRf -** Enter to set parameters for the Left Front/Right Front upmixer:

    - **Enable Center -** 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**.

    - **Enable 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**.

- **Enable LFE -** A checkmark indicates that an LFE channel is being created to produce the 0.1 channel of the upmixer. Default is **Enabled**.

- **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.!00% 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%.**

- **Use Center Width for 5.1 (On/Off) -** Allows discrete center channel audio to also be spread across the front channels by the percentage set by Center Width. Conceived by our friends David and Darren at Ascent/Discovery UK, this mode is useful for content that has been produced with only dialogue in the center channel (allows for alternate languages). Note that the setting in this menu and in the Upmix LfRf menu operate in parallel. Default is Off.

- **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 **-10dB.**

4.2.2   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 form 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.

- **LfRf Upmix: Off -** Indicates the current state of the upmixer. Used to show whether the Automatic Threshold has commanded the upmixer to engage or disengage.

- **Discrete Threshold -** 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)

- **Upmix -> Discrete Xfade -** Sets the timing of the transition from Auto Up-mix 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 is **Slow**.

- **Discrete -> Upmix Xfade -** Sets the timing of 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.

- **Bass Enhancement (On/Off)-** Produces an LFE channel form the 80Hz low pass filtered sum of the discrete Left, Center, and Right inputs. Use with care as in many cases consumer subwoofers are mis-aligned.

- **Level (0dB - 24dB) -** Adjusts the output level of the created LFE signal. Default is -10, but please adjust to taste. Some installations have found lower values such as -18 dB to be a better match for their viewers.

4.2.3   Local Input (AERO.air (5.1) Only)

When the AERO.air is in 5.1+Local Program Config (under I/O Menu), Main In 7/8 can be used to replace or combine with the audio applied to Main In 1-6. In addition to up-mixing as described above, voiceover with automatic ducking can be performed. Called VOX mode, it is useful where no contact closures are available to enable switching. The settings are as follows:

- **Destination:** Selects destination of Program 1 (Ch. 1-6) or Program 2 (chan 7/8) for the insertion of Local audio.

- **Input Mode:** Selects between Left (audio from Channel 7), Right (audio from Channel 8), or Stereo (audio from Channel 7/8).

- **State:** Selects the current mode of the Local Input:

  - Off(VOX) - Main program will fade to level selected below under Duck Level when Local Audio is present

  - Over - Ducks main program as set below, Local mixed over

  - Replace - Local input replaces Main input

- **VOX Tail:** Sets the delay before ducking of Main Program audio stops after Local

Program audio is removed. Default is 500msec

- **Duck Level:** Sets the attenuation level in dB of the Main Program when duck is active

- **X-Fade:** Sets the speed at which the ducking is performed. Default is Quick

- **Enable Upmix:** Allows local audio to be upmixed to 5.1

4.2.4   CrowdControl (Option) for 7/8 Output

Most television sets today have so-called hyper surround modes which have the intended effect of boosting certain L-R signals in both linear and non-linear fashions to produce a widening of the stereo image. Unfortunately, this can also have the unintended effect of overpowering certain components present in the L+R signal, namely the dialogue especially when the original content is "phase rich" such as with surround programs. Crowd control is adaptive pre-correction that provides the minimal amount of processing necessary to protect dialogue and minimizes audible effects on surround encoded material.

- **Enable -** Turns the process on or off (somewhat like the name would suggest)

- **Clarity Enhance (0-100) -** Determines what percent of certain components of the L+R are boosted with respect to others, specifically surrounding the dialogue region. Adjust to bring out dialogue that is lost in the background. Default is **40**.

- **Inv Freq Comp (0-100) -** Determines the percentage of pre-correction applied to the input signal to "un-do" some of the linear and non-linear processing that will be applied by many stereo consumer television receivers. Default is **51**.

- **Max Sur Red (0-12 dB) -** Sets the static level at which certain components of the L-R signal are reduced with respect to other components. Provides a limit to the amount of dynamic correction that will be applied to a given audio signal. Default is **9 dB.**

4.2.5   7/8 Source (AERO.air (5.1) Only)

Allows selection of audio on 7/8 input as follows:

- **.7/8 -** Stereo audio applied to 7/8 input is passed as stereo

- **7 Only -** Input channel 7 is copied to 7 and 8 paths as mono

- **8 Only -** Input channel 8 is copied to 7 and 8 paths as mono

4.3   Modify Processing Menus

Modify Primary, Secondary, Tertiary (and Quaternary if in 2+2+2+2) Programs here. Note that in the case of the AERO.air (DTV) there is only one primary program.

- allows adjustment of all processing parameters - the "Adjust" section is covered in great detail in Chapter 5 of this manual.

- saves and recalls processing presets - All configuration begins with a factory preset but these can be modified, stored, renamed, and recalled. All processing core parameters are saved in the preset.

Saving a preset is a simple procedure. Adjusting any parameter in the **Modify Processing** menus appends an *M* to the name of the original preset to indicate that it has been modified but not yet saved. Figure 4-2 shows the **Save Preset** menu.

Figure 4-2 Recall Preset menu (left) and Save Preset menu (right)

The **Save Preset** screen allows you to keep the existing name or rename and save using the arrow keys and the rotary encoder. To rename, press the Up arrow to highlight the current name, then press the Right (Next) arrow to enter edit mode (indicated by the <> symbol), then use the Left and Right arrows along with the rotary encoder to change the preset name. When you are satisfied with the name, press the Down arrow to highlight the Save selection, then press the right arrow to save. If you attempt to write over a previously saved preset name, you will be prompted to confirm that you wish to overwrite a stored preset.

CAUTION:The system allows overwriting factory presets, and you will be warned!

4.4   System Info Menu

The **System Info** menu provides real-time statistics regarding the current state of the AERO.air. It is a quick way to judge overall system function and status. Below is the system info screen and following are descriptions of the indicators.

:
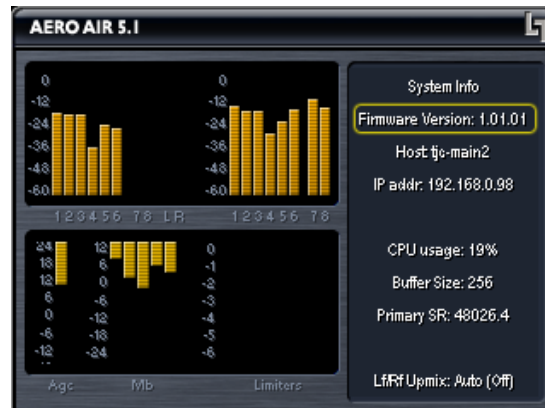
Figure 4-3 System Info menu

- **Software Version** - System software version, currently V1.17.14 2U

- **CN1000 Ver** - Micro controller version, currently V1.3

- **FPGA Ver** - Firmware version, currently 0x10000110

- **CN961** - Dolby encoder card, if present will show Active

- **Host** – Unit name (normally AMX200XXX where XXX are from the unit S/N)

- **IP Addr** – 127.0.0.1 (dynamically allocated, so this will change IF connected); unique MAC address is directly beneath the IP address.

- **CPU Usage** – Normally around 26%

- **Buffer Size –** The size of the audio buffer in samples.

- **Primary SR –** Average Sample Rate, nominally 48 kHz

- **HP Output SR –** Average Sample Rate for headphone path, nominally 48 kHz

- **LfRf Upmix** – Indicates whether upmixing is currently on or off

- **SDI Video Mode** – Indicates presence and type of SDI signal applied:

    - HD-SDI or SD-SDI

    - SMPTE format and description: 274M 1080i

- **Metadata** – Indicates current status of incoming metadata:

    - Metadata OK/Failed

    - Primary Acmod: Program 1 audio coding mode such as 3/2L

    - Primary Dialnorm: Program 1 dialnorm, such as 20

    - Secondary Acmod: Program 2 audio coding mode such as 2/0

    - Secondary Dialnorm: Program 2 dialnorm, such as 20

- **GPI Status: Enabled/Disabled**

    - Individual GPI Input Status (1-8 + Metadata GPI): HHHH HHHH L

- **Options -** Displays installed options such as CN561 (Dolby encoder), SDI, SIMM (Dolby decoder), etc...

4.5   System Options Menu

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

4.5.1   Master Bypass

Relay bypass of ALL inputs to their respective outputs

4.5.2   I/O Options

This is the menu for general system, audio, and clock setup:

- **PGM Cfg (AERO.air (5.1) Only)** – Select between the following Program Configurations (see the end of this chapter for block diagrams of each mode):

    - 5.1 + 2 - Main audio on inputs 1-6, separate two channel program on 7/8, and EAS input on 9/10 (digital or analog).

    - 5.1 + Loc - Main audio on inputs 1-6, local two channel local or voiceover audio on 7/8 and EAS input on 9/10 (digital or analog).

- 5.1 + 2 + Local - Main audio on inputs 1-6, separate two channel program on 7/8, and local or voiceover input on 9/10 (digital or analog). Use this mode for 5.1+2+2 operation. EAS remains available via the analog input if not used for Local audio.

- 2 + 2 + 2 + 2 - Four separate audio programs on 1/2, 3/4, 5/6, and 7/8, and EAS input on 9/10 (digital or analog).

- **7/8 Src** – Selects source for 7/8 output. Choices are 1-6 In (LtRt), 1-6 In (LoRo) or 7/8 In. In all cases, the audio sources are pre processing.

- **Downmix -> 7/8 (AERO.air (5.1) Only)** – Select to replace the 7/8 output (and the audio encoded by Dolby Digital (AC-3) Encoder B) with a downmixed version of the main program. Active in 5.1+2 mode only. Can also be triggered automatically or via GPI

NOTE: *Downmix ->7/8 can be invoked automatically or via GPI. This is particularly useful when channels 7/8 are supporting SAP or Auxiliary audio that might not always be present. When not present, a downmixed version of the main audio can be inserted automatically or on GPI command providing audio continuity on the 7/8 Output. See Input Switching for setup of Auto Downmix mode.*

- **Extra Delay (Ch 1-2, 3-4, 5-6, 7-8, 9-10) -** Allows up to 300 milliseconds of additional delay to be added to each OUTPUT pair for plant timing purposes. Default of 0 msec is equal to 33 msec of total input to output latency.

- **Gain 7-8** – Allows output level of Main Out 7/8 to be attenuated over the range of 0dB to -31dB. This is useful for calibrating the downmix output to match the 5.1 path, especially in systems where the 5.1 is carried via AC-3 and the downmix is carried via MPEG audio. Since AC-3 is normalized to -31 after decode, the MPEG path may require additional attenuation to match.

- **Audio Inputs** – 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)

- **Dolby E Bypass** – When Dolby Decode option is installed, enabling Dolby E Bypass inserts detection and bypass of PCM audio around the Dolby E decoder when the Dolby E input signal turns to PCM. This precisely emulates the operation of a Dolby DP572 decoder so that transitions are free of clicks or pops and enables an "Alternate PCM" routing to take place. See the end of this chapter for a detailed description of Dolby E features.

- **Comp Delay PCM** – When the Dolby encoder option is installed, enabling Compensating Delay for PCM channels adds the exact delay caused by Dolby encoding to the PCM channels (1-10) to that encoded Dolby Digital or Dolby Digital Plus and PCM outputs are in time. Currently, this delay is 137 msec.

- **Enc B Out** – The fifth BNC connector on the rear panel can be set to output Encoder B (Dolby Digital or Dolby Digital Plus), 9/10 PCM audio, or a down-mix of the main program (post delay and post processing). Default is Enc B Out = Enc B.

- **SDI Embed** – Select to enable SDI embedding.

- **Clock Ref** – Select between SDI, AES In 1, AES Ref, or Internal 48 kHz.

NOTE: *When SDI Embed is enabled, Clock Ref will automatically be forced to SDI to maintain proper synchronization.*

- **SDI Outputs** – 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) ...

    - **Out 15/16 = SDI 15/16** (AES 3-10, Enc A, Enc B, SDI 1/2 - 15/16)

- **SIMM Source** – Main Input for the optional Dolby decoder SIMM card. Any AES or SDI input can be selected to feed audio to the SIMM card. Please see the end of this chapter for more detailed information regarding the Dolby decode option.

- **SIMM Bypass Source** – Alternate PCM source for PCM Bypass operation. Please see the end of this chapter for more detailed information regarding the Dolby decode option.

- **Test Tones** – See below

- **EAS** – Select the destination of the EAS/Aux audio insertion: Pgm 1, Pgm 2, or All programs. The EAS or Auxiliary audio will completely replace all audio in the chosen destination program(s).

- **Lock Headphones** - Selecting this menu item locks users out of the Headphone adjustment menus, maintaining the last settings entered. This is useful to prevent accidental changes.

- **Clock Ref** – Select between SDI, AES In 1, AES Ref, or Internal 48 kHz (this is a duplicate of the menu near SDI settings, included for convenience).

- **Metadata (status)** – Indicates presence or failure of input metadata (RS-485 or VANC). Presence will be shown as the audio coding mode of the applied program (i.e. 3/2 or 2/0).

- **Metadata Source (RS-485/VANC/SIMM) -** Selects between RS-485 (serial 9-pin) metadata, metadata extracted from the vertical ancillary VANC) space of an applied HD-SDI signal, or metadata from the optional Dolby Decoder SIMM if installed.

NOTE: *If SIMM is chosen as the metadata source, it will also act as the metadata source for the Dolby Encoder option, taking full control of the audio coding mode, dialnorm, DRC, and other metadata values. Care must be exercised if metadata is also used to control upmixing as upmixed audio must be encoded in 3/2L mode- mismatch may occur.*

- **VANC DID: -** Sets the Data ID for metadata extraction. Default is 0x45.

- **VANC SDID: -** Sets the secondary Data ID for metadata extraction. Default is 0x01.

- **VANC Extract Line -** Sets the video line for metadata extraction. Default is Auto.

- **VANC Async -** Enable for SMPTE 2020M "B" mode (i.e. CBS Metadata)

4.5.3   Test Tones

Test Tones – Multiple setup tones are available to calibrate the AERO.air and other downstream equipment. The **Level** control is self-explanatory, and is set at the standard –20 dBFS reference level and adjusts the overall level of the test signals. **On/Off** turns the selected test signal on or off while **Cycle** determines which channels will reproduce the selected test signal. It can be adjusted to cycle through all channels, all channels of just the main program (default), and individual channels.The available test signals are:

- Sine – 1 kHz sine wave

- Voice – Voice ID of all selected channels

- Pink Noise – Equal energy-per-octave noise

- White Noise – Equal level-per-octave noise

- Brown Noise – Low-pass-filtered noise

- Brown Nose – Try it (because broadcast audio is a funny business)

4.5.4   Record Audio to USB

Allows for capture of audio 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.

4.5.5   Nielsen Watermark (Option)

Linear Acoustic has been working with the Nielsen Company to enable the inclusion of their audience measurement technology in the right place between audio control and encoding. The result is that the AERO.air is now optionally capable of inserting a Nielsen Watermark for audience measurement into any or all of the five output channel pairs. The unit can be configured with up to five assignable watermark encoders, each capable of handling two channel (mono or stereo) or 5.1 channel sources.

The Nielsen Watermark encoder inserts data into the audio signal in a manner that keeps the it hidden, or masked, behind the audio. If there is little or no audio present, there is little or no watermark inserted. This results in an impressively well hidden data signal that is robust enough to be detected by Nielsen Watermark decoders but is otherwise undetectable by viewers under normal listening conditions.

Each watermark encoder can be configured to apply the watermark to any single pair, or to three pairs for a 5.1 channel service.

- Audio Output – Selects which pair or pairs of audio channels will have the Nielsen Watermark inserted. Any single pair can be selected, or a group of three pairs for channels 1-6 can be selected for 5.1 channel services.

WARNING: to maintain proper phase coherence, a 5.1 channel service MUST be watermarked by a single encoder. DO NOT use three separate encoders!

- SID – Station ID, uniquely assigned by Nielsen Company to each station or service. A Test SID of 9001 is present by default and uses check digits (described below) of OA.

- Check Digits – Provided by Nielsen, the check digits are a type of CRC that ensure the

SID is valid. Characters are uppercase alphabetic and are unique to each SID. Incorrect check digits or SID will prevent watermark from being enabled.

- Distro – "Distribution" indicator, selects between Final Distribution or Pgm Content. The normal mode for a television station or cable network is Final Distribution, and this is the default setting.

- Ins – Insertion Mode, Refrain/Overwrite. Default is Overwrite.

4.5.6   Pgm Cfg Automation (AERO.air (5.1) Only)

Program Configuration Automation allows for a major mode change of the AERO.air to be accomplished via a simple control. Program Configuration sets the overall processing signal flow through the AERO.air. For example, 5.1+2 is two programs, the first with audio on channels 1-6 and the second with audio on channels 7/8. A program config of 5.1+2+Local is three programs with the first program on 1-6, the second on 7/8, and the third or tertiary on 9/10. A program config of 2+2+2+2 is four programs with the first program on 1/2, the second on 3/4, the third or tertiary on 5/6, and the fourth or quaternary on 7/8.

Within this menu, two different configurations can be defined, one for each state of the Pgm Cfg GPI control. The five settings for each are as follows:

- Program Config: Select between the available channel configurations for the AERO.air (5.1): 5.1+2, 5.1+Local, 5.1+2+Local, or 2+2+2+2.

- Preset 1-4: Select the processing preset for each of the four possible programs.

4.5.7   GPI Setup

This menu allows configuration of the 9 GPI inputs, comprised of GPI A-H and GPI M (Metadata). Each can be set to provide a specific function that will be active while either the GPI pin (A to H) which are internally held High are externally pulled and held in a low state, or when the metadata input is active (GPI M). Note that each GPI must have a different function, meaning that if GPI A is set to Upmix LfRf, then none of the other GPIs can be set for any of the Upmix LfRf functions.

When GPI is enabled (the **Enable GPI** menu item is highlighted), the unit waits for an external command via the GPI/O port and nothing else can be changed. To make a change, disable GPI by de-selecting the menu item.

The available functions include the following:

- Disable

- Upmix Lf/Rf: Off/On; On/Off - Creates a 5.1 channel upmix (i.e. upconversion) from

the Channel 1/2 (Lf/Rf) input. Off/On causes upmixing to be engaged upon GPI command, while On/Off is the inverse state where upmixing is on by default and off upon GPI command.

- Upmix Off/Auto; Auto/Off; Auto/On - 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. Off/Auto and Auto/Off are self explanatory, but Auto/On is very useful in situations where a user knows that certain content is two channels and desires a full upmix and no feed-through of the remaining channels.

- Local **(AERO.air (5.1) Only)**: Off/Mix; Off/Replace; Mix/Replace - Mix allows for a standard manual voiceover as defined in the Mode Switches menu, while Replace simply replaces the main audio with the local signal.

- EAS Mode: Off/On; On/Off - Mutes main program audio and outputs to Left front/Right front the signal present on the EAS Input. Off/On defaults to EAS Mode off and enters EAS Mode upon GPI command while On/Off is the inverse state where EAS Mode is on by default and off upon GPI command.

- Downmix 7/8 Off/On; On/Off **(AERO.air (5.1) Only)** - Active only in 5.1+2 mode, allows downmix of primary program (channels 1-6) to replace the +2 audio signal on the channel 7/8 output thus providing audio continuity on an SAP/Aux channel when 7/8 input is not present.

- Preset selection (Primary and for AERO.air (5.1) also Secondary, Tertiary, Quaternary) - Allows GPI to select between a main and backup preset if configured in the Adjust Primary -> Automation menu. Also present in the Adjust Secondary, Tertiary, and Quaternary Automation menus. When GPI is Hi (i.e. no closure), the main preset is active, and when GPI is asserted low (i.e. closure), the backup preset is active.

- 7/8 Mode **(AERO.air (5.1) Only)** - Allows automation of 7/8 audio inputs to support SAp and DVS operation. Selections include: Stereo/Mono 7, Mono 7/Mono 8, or Stereo/Mono 8.

- Pgm Cfg Automation - Allows GPI control of Program Configuration and processing presets for each of the programs via a pre-set table of values. Please see details above under the Pgm Cfg Automation description.

Metadata->GPI M State – A Dolby Metadata compliant signal applied to the Metadata In connector on the rear panel of the AERO.air can also be used to drive the above functions. Metadata control of the unit is implemented as GPI M. Normally, metadata is examined to determine if the main program is two channels, and if it is then it can trigger a GPI function. There is also a setting to tell the unit what to do in case of metadata failure.

The adjustments are as follows:

- 2 Channel = GPI M On (Off); Default is On

    If two channel mode is detected for program one of the incoming metadata stream, then GPI M is activated (GPI C function can be set as described above). If two channel mode is not detected, then GPI M is de-activated.

- Metadata Fail = GPI M (On) Off; Default is Off

    Determines what to do should the incoming metadata stream fail. Default is for GPI M to be de-activated. If the incoming metadata stream is valid, then no change to GPI M is made (i.e. normal GPI M functions continue to operate as controlled by metadata). For example, this will allow the unit to stop upmixing if the metadata fails, thereby passing all audio channels straight through. In the case of the AERO.air (5.1) it can also be used to activate either the local (7/8) or Voiceover inputs in case of failure. Metadata Fail would be set to GPI M = On, and GPI M would be set to Local Off/ Replace or Voiceover Off/Replace.

WARNING: Setting the Metadata Fail to activate upmixing is dangerous. If a 5.1 channel program is present and metadata fails, the dialog will be muted. Use Auto!

Current Metadata: Shows Metadata Program Config or Fail.

4.5.8   Time and Date Settings

Allows configuration of the AERO.air 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 unit is connected, configuration of the facilities Ethernet router may be necessary

NOTE: *The AERO.air currently 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.*

4.5.9   Preferences:

- **Change Password** – The password is either entered here for the first time or changed. Use the rotary knob and the right arrow key to enter your password, then select **Save.** You will be prompted several times, and then you will be reminded not to lose the password. *Please heed the warning!* If you do happen to lose it, please call us to assist recovery without losing presets. If you wish to remove the password, simply enter the old password, then enter nothing for the new one and select **Save**.

WARNING: If storing a password for the first time, it is HIGHLY recommended to "Save Settings To USB" as described on the next page. Then, store the drive in a safe place in case the password is lost.

- **LCD Shutoff** – Select the time-out for the OLED display. This setting is currently ignored.

- **Reinitialize** – Fast reset of the clocks and system software

- **Save Settings Now –** To minimize extra wear on memory, changes to the system are automatically saved after about 5 minutes of no activity. However, it is sometimes useful to be able to save immediately. Selecting this function will do so, and will remind the user that it takes a few seconds even after the function has been performed.

- **Save Settings To USB –** Allows unit settings to be stored to a factory supplied or factory approved USB device connected to the front panel USB connector. When this device is installed in an AERO.air unit and the system is power cycled, a startup message will prompt the user to allow restoring all settings or presets from USB.

- **Update Software –** When a factory approved USB device (such as the one included with the unit) has an update copied to its root directory and is inserted into an AERO.air, selecting this option will start the software update process. Please note that this process will cause interruption to the audio, possible for several minutes depending on the type of update. This is the method used to update software, firmware and microcontroller versions.

WARNING: Selecting Update Software will cause audio to be interrupted for some period of time and can extend for several minutes depending on the update. You will be prompted before the update process begins.

- **Power Cycle –** Slow reset of the entire system. Select this to perform a full system reboot and re-initialize; you are returned to the main menu.
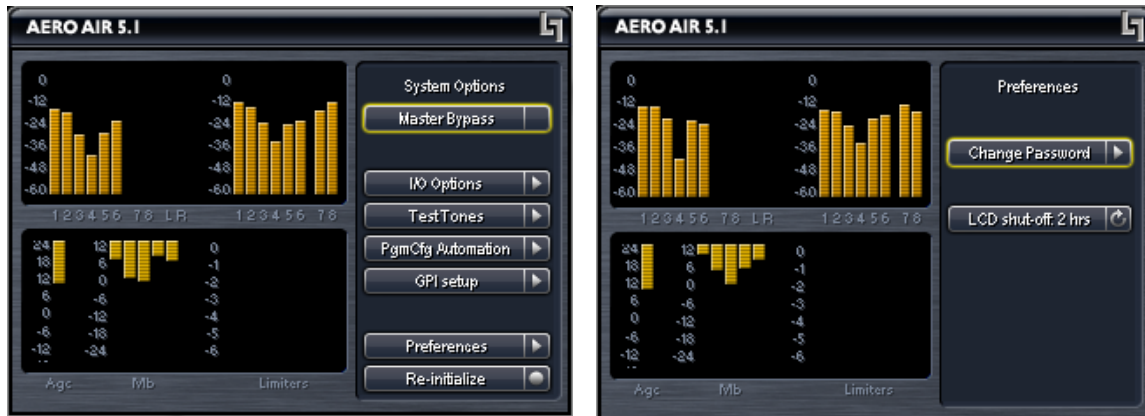
Figure 4-4 System Options menu (left) and Preferences menu (right).

4.5.10 Program Configuration Diagrams

The following pages contain block diagrams depicting signal flow in each of the different program configurations. While at first this may seem somewhat complex, it demonstrates the incredible flexibility and values of the AERO.air platform. Almost any audio application in a modern digital television facility can be accommodated.

Please note that SDI, encoding, and metadata options are not explicitly shown here for the sake of clarity. However, it can be reasonably assumed that an input channel can come from either AES or any of the embedded SDI audio channels, and the encoders are directly sourced from Channels 1-8. Please see Chapter 1 and the general system block diagram for more information about the capabilities of the AERO.air platform.

The first diagram shows the entry level AERO.air (DTV) which has a single processing core which includes UPMAX and UPMAX II upmixing and downmixing. The remaining diagrams show all of the possibilities of the fully-featured AERO.air (5.1) version.

NOTE: *An AERO.air (DTV) can be optionally upgraded to a full AERO.air (5.1) easily and at any time. Please contact the factory for details.*

5.1   **+**   **(EAS)**

1/2   **3/4**   **5/6**   **EAS/Aux
9/10**

**UPMAX
AutoMAX-II**

AEROMAX 5.1
Processing Core
(Primary)

Norm/EAS

6

**Downmix
Encoder
(LtRT)**

**Downmix
Encoder
(LtRT/LoRo)**

RFI

CrowdControl

1/2   **3/4**   **5/6**   **Local
(7/8)**

**9/10
Monitor
LtRt/LoRo**

L   **R**

**Headphone
Output
(Via FP 1/4")**

© Linear Acoustic Inc.
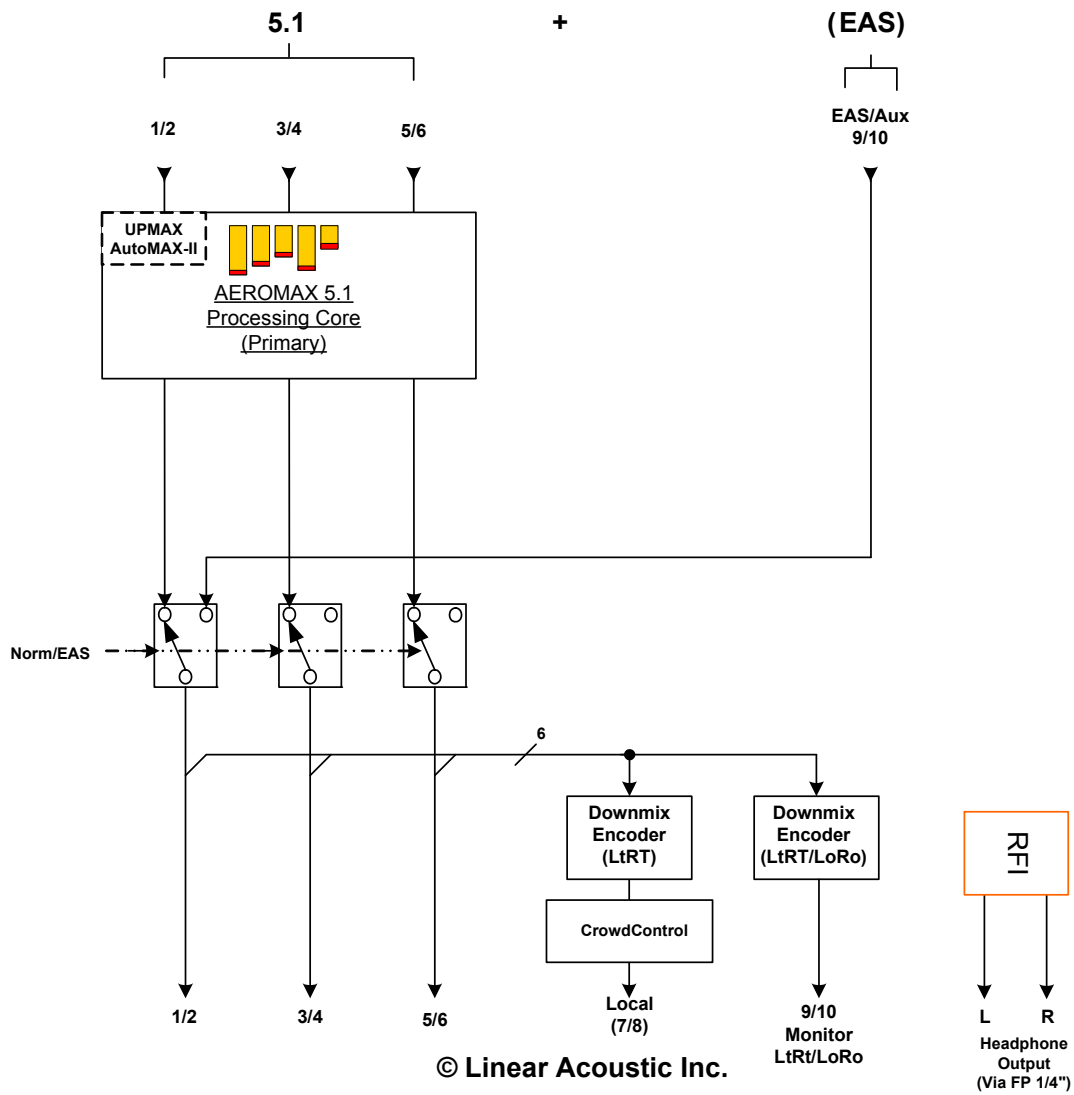
Figure 4-5 AERO.air (DTV)

The entry level AERO.air (DTV) can accept up to 6 channels of audio plus EAS emergency audio and will create a 5.1 channel main output plus a two channel downmix output with CrowdControl dialogue protection processing.
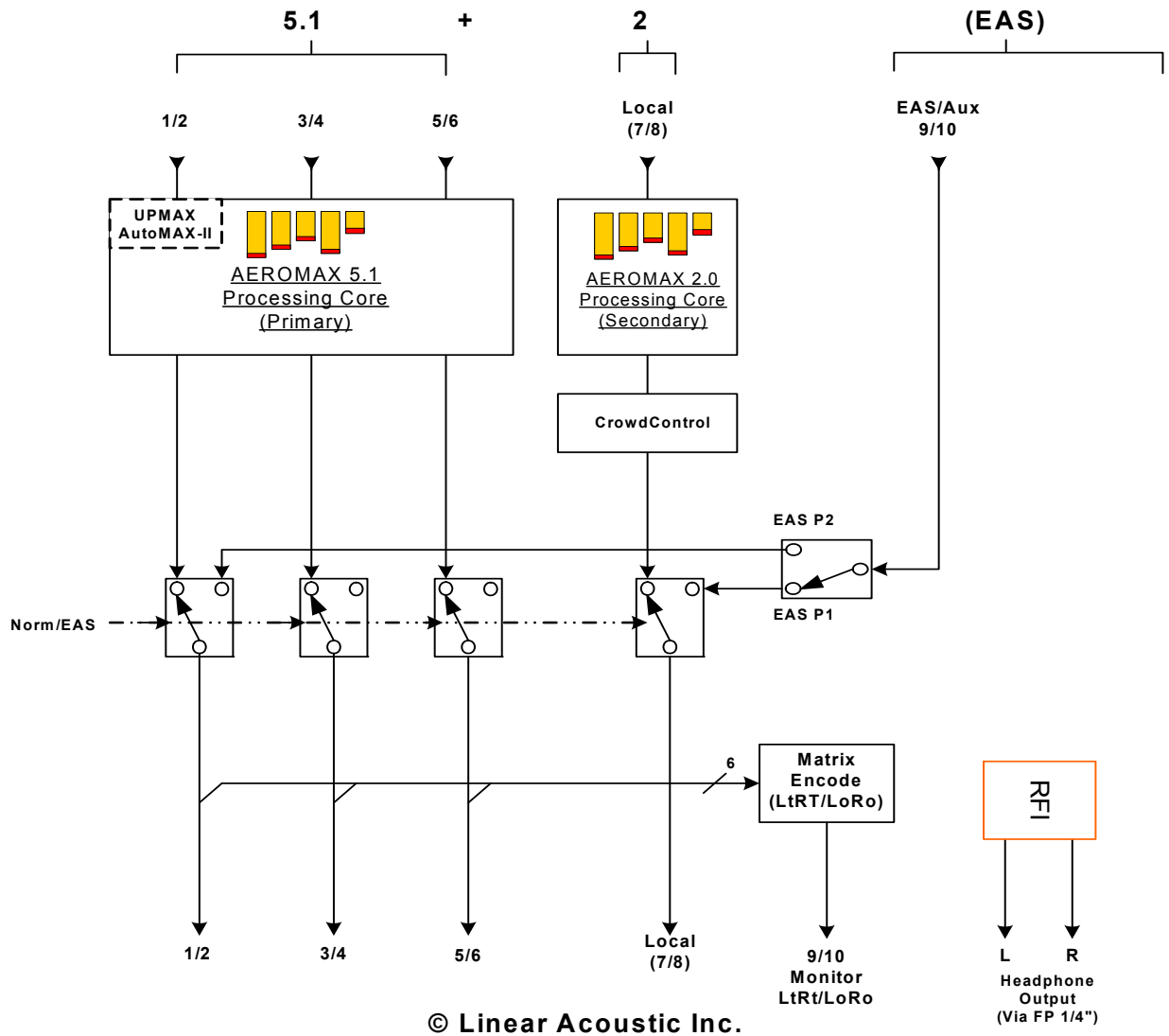
Figure 4-6 AERO.air (5.1) in 5.1 + 2 Program Configuration

This is one of the most popular modes the AERO.air is used in, allowing a 5.1 channel main program plus a secondary related or unrelated stereo program to be processed independently. EAS audio can be inserted after processing to either or both programs via GPI control.
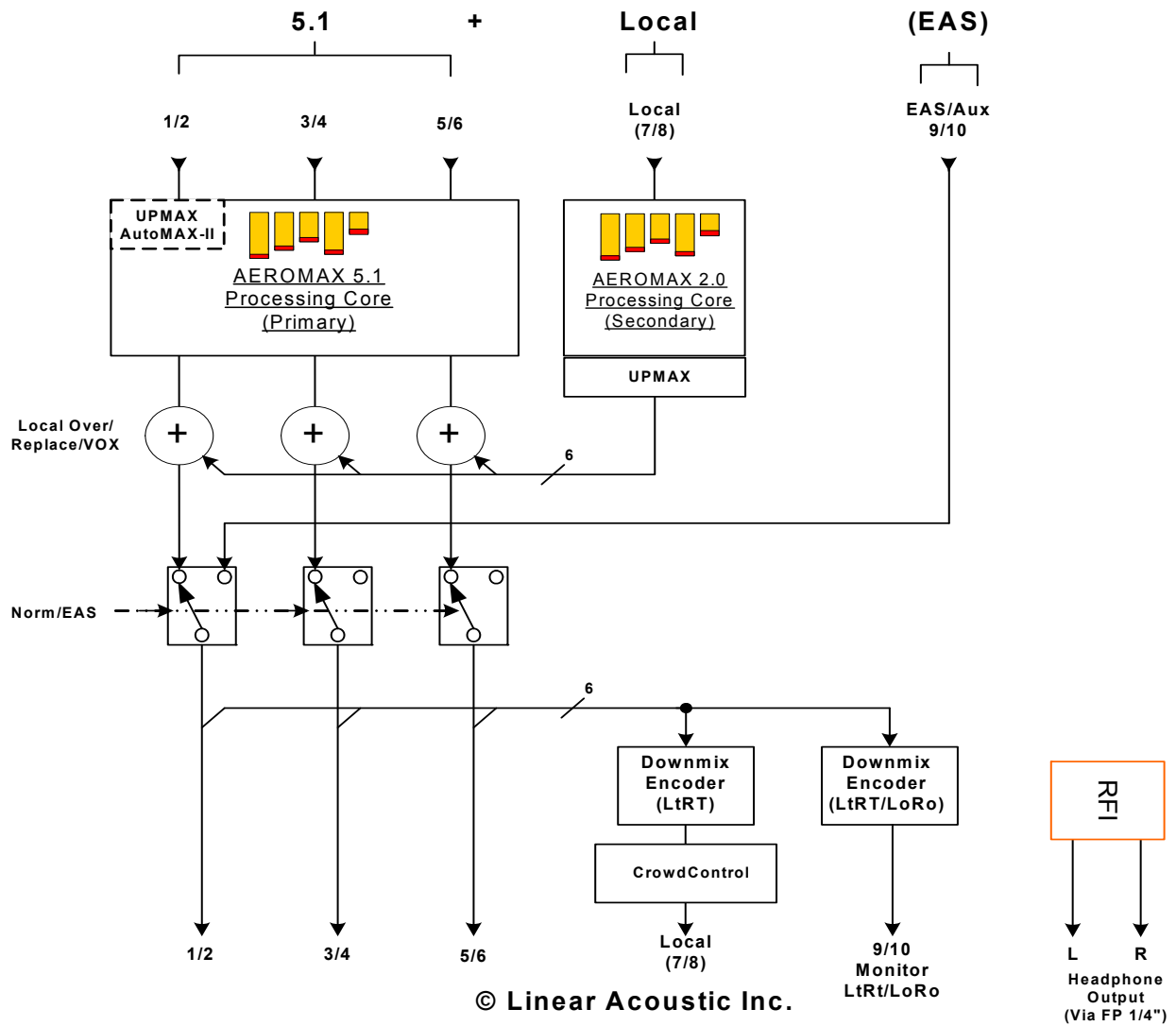
Figure 4-7 AERO.air (5.1) in 5.1 + Local Program Configuration

This is another commonly used mode and it allows a 5.1 channel main feed, such as from a network, to be combined with a stereo or dual channel mono local feed. The AERO.air can upmix the local audio to 5.1 channels if desired and can crossfade between the main and local programs on command. EAS audio can be inserted after processing via GPI command.

Figure 4-8 AERO.air (5.1) in 5.1 + 2 + Local Program Configuration

This mode is useful for handling a 5.1 channel main program plus and unrelated +2 program (such as a weather feed or an alternate stereo feed) and having an analog or digital Local input on 9/10 for processing, upmixing, and inserting voice-overs or other sources.

This mode can also be used to support so-called 5.1+2+2 mode if local voiceover functions are simply not used. By selecting Program 3 in the Headphone menu (selects source for 9/10 analog and digital outputs). Once selected, Lock Headphones can be chosen in the I/O menu to prevent on-air changes.
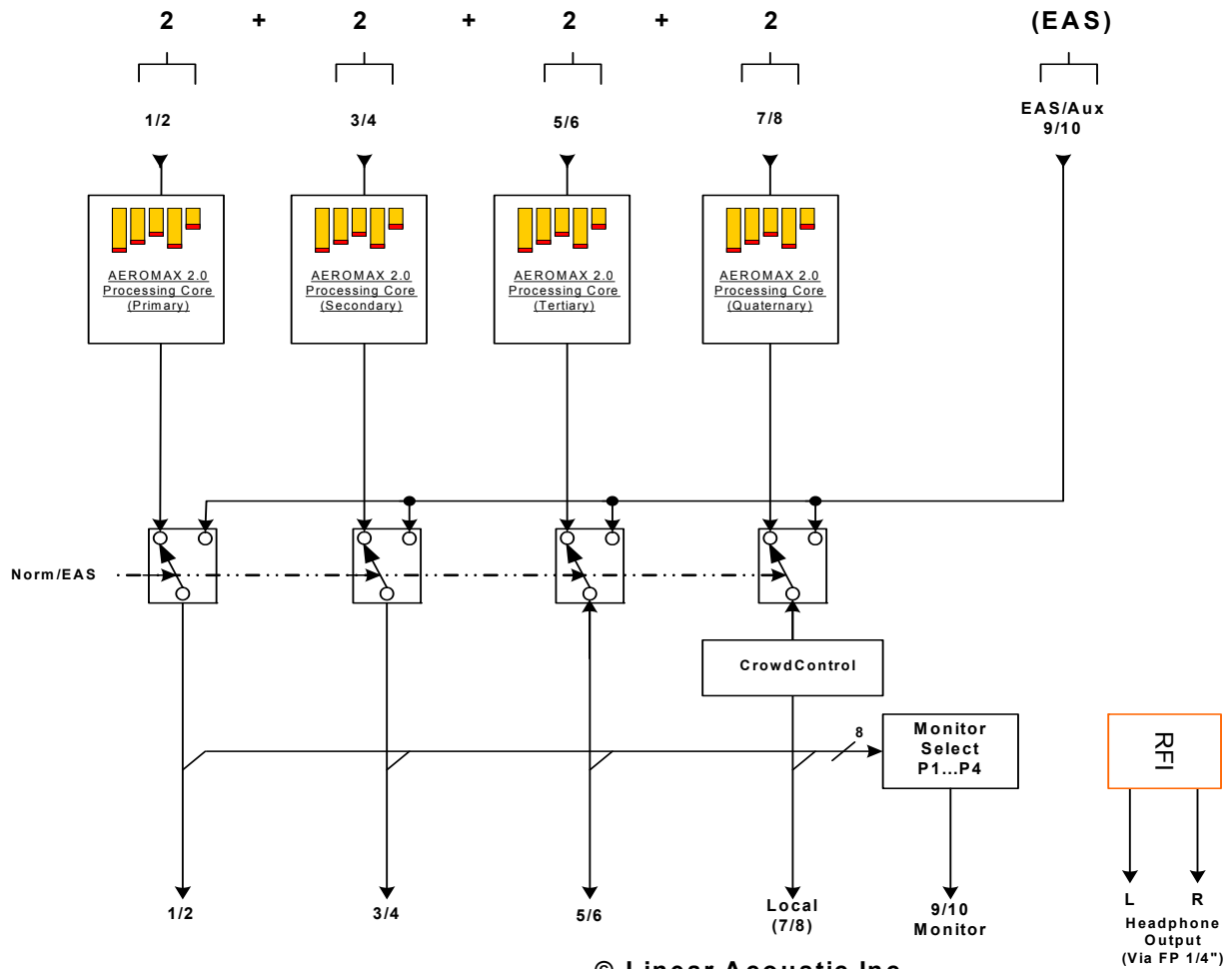
Figure 4-9 AERO.air (5.1) in 2 + 2 + 2 + 2 Program Configuration

This mode allows for four separate stereo programs to be processed completely independently. Each program will have its own dedicated processing engine, and the monitor outputs can select any one of the four programs.

4.6   Dolby Decode Option

Dolby E, Dolby Digital (AC-3), and Dolby Digital Plus (E-AC-3) input signals can be decoded to PCM for loudness control, upmixing, and output as either PCM and/or re-encoded to Dolby Digital or Dolby Digital Plus. This option is primarily intended for decoding Dolby E and its associated metadata.

Implementation of the Dolby decoder matches that of a Dolby DP572. Unlike other systems, transitions between PCM and Dolby E are handled gracefully and without artifact or noise. Users should note that there are two inputs to the Dolby decoder: Main and Bypass. This enables handy re-routing of signals as they transition between Dolby E and PCM, functionality particularly useful for managing 5.1 channel audio and SAP tracks.

The tables below show the operation.

Table 4-1 SIMM with Dolby E Input

Input	Signal	SIMM Output
SIMM Source	3/4 Dolby E	1/2 = Main 5.1 Lf/Rf
		3/4= Main 5.1 C/LFE
		5/6 = Main 5.1 Ls/Rs
		7/8 = SAP
SIMM Bypass	1/2 Main LtRt	-

Note that when the SIMM is decoding Dolby E, the main program is on channels 1-6 and the alternate language SAP audio is on 7/8.

Table 4-2 SIMM in PCM Bypass

Input	Signal	SIMM Output
SIMM Source	3/4 SAP	1/2 = Main LtRt
		3/4= -
		5/6 = -
		7/8 = SAP
SIMM Bypass	1/2 Main LtRt	-

Note that when the SIMM input becomes PCM, the system automatically maintains main program audio on channels 1/2 and the SAP is re-routed to 7/8 and continuity is maintained.

Linear Acoustic AERO.air Menu Tree (Page 1)

(SW Version 1.18.01 2U and higher; Shown with Nielsen and Dolby Digital Encoding Options Installed)

See Next Page

Top row / AERO.air System Options branch:
- I/O Options
- Test Tones
- Record Audio to USB
- Nielsen Watermark *
- PgmCfg Automation
- GPI Setup
- Time/Date Settings

- AERO.air System Options
- Master Bypass
- Metadata Bypass

- Preferences
- Re-Initialize
- Save Settings Now
- Save Settings to USB
- Update Software
- Power Cycle

Metadata branch:
- Metadata Source: PGM1 (1-8)
- Metadata Apply Dialnorm: **V**
- Metadata Global def. dialnorm: 20
- Metadata Metadata: OK/Fail
- Metadata Acmod: 3/2
- Metadata Dialnorm: 27

Processing Core Adjust:
- Adjust Noise Reduction
- Adjust Parametric Eq
- Adjust Input AGC
- Adjust Multiband AGC
- Adjust Multiband Limiters
- Adjust Multiband Eq
- Adjust Final Stage

- Final Limit Drive: -3dB
- Master Level: -11dB
- Current Output: -24 LKFS
- Utility Loudness Meter

Modify Primary branch:
- AERO.air Modify Primary ( )
- Modify Primary Recall Preset
- Modify Primary Save Preset
- Modify Primary Metadata
- Modify Primary Adjust
- Headphone Output Level: -6dB Source: PGM1 (1-8) Downmix: LoRo (LtRt)

Dolby Digital Encoder (OPTION):
- Dolby Digital Encoder (OPTION)
- Bypass AC-3 ->

Encoder 1 (Ch 1-6):
- Data Rate: 384k
- Coding Mode: 3/2L
- Line Mode: None
- RF Mode: None
- Dialnorm: 27

Encoder 2 (Ch 7-8):
- Data Rate: 192k
- Coding Mode: 2/0
- Line Mode: None
- RF Mode: None
- Dialnorm: 27

Present in AERO.air (5.1) Version ONLY

AERO.air System Info:
- AERO.air System Info
- Firmware Vers.: 0.505 CN1000 ver: 0.8 FPGA ver: 0x10000008 CN561: Active ...

Upmix Local:
- * Upmix Local
- Enable: **V**
- Enable Center: **V**
- Enable Surround: **V**
- Enable LFE: **V**
- Center Width: 33 % Use Ctr Width for 5.1
- LFE Level: -10dB

Crowd Control:
- Crowd Control
- Enable: **V**
- Clarity Enh.: 33
- Inv Freq Comp: 42
- Max Sur Red: 12.00dB

AERO.air Mode Switches:
- AERO.air Mode Switches
- LtRt Upmix* On/Auto/Off ->
- Upmix LtRf

- *Upmix Local (in 5.1+Local mode)
- Bass Enhancement (On/Off)
- Level: -6.00dB
- CrowdControl ->
- * Local | Voiceover

Local Voiceover:
- Input Mode: Stereo (Left/Right/Stereo)
- State: Off (VOX) (Over/Replace/Off)
- VOX Tail: 500ms (Off-3sec)
- Duck Level: -12.0 dB (Full-0dB)
- X-Fade: Quick
- Enable Upmix (On/Off)
- * 7/8 Source: 7/8 (7 only/8 only)
- Auto Downmix ->
- Headphone Output Level: -6dB Source: PGM1 (1-8) Downmix: LoRo (LtRt)

AERO.air Main Menu (Top):
- AERO.air Main Menu (Top)

*When GP active, this is upmix status, when GP not active, -> controls upmix.

Upmix LtRf:
- Upmix LtRf
- Upmix: Off/On/Auto
- Enable Center: **V**
- Enable Surround: **V**
- Enable LFE: **V**
- Center Width: 33 % Use Ctr Width for 5.1
- LFE Level: -10dB
- LtRf Upmix: Off
- Discrete Thresh: -75dB
- Upmix->Discrete Xfade: Slow
- Discrete->Upmix Xfade: Very Quick

- Xfade Quick
- Holdoff 2.0 sec

Linear Acoustic AERO.air Menu Tree (Page 2)

(S/W Version 1.18.01 2U and higher; Shown with Nielsen and Dolby Digital Encoding Options Installed)

Time/Date Settings
- YYYY-MM-DD HH:MM:SS
- Day
- Time Zone
  - Time Zone
    - (GMT)
    - Greenwich Mean Time Dublin, Edinburgh Lisbon, London
    - Daylight Savings **v**
    - Apply (Restart Req'd)
- NTP Settings
  - Enable **v**
  - Server | Addresses
    - 0 pool ntp org
    - 1 pool ntp org
    - 2 pool ntp org
    - 3 pool ntp org
    - 4 pool ntp org
    - Apply
- NTP Enabled
- Time: HH:MM:SS

GPI Setup
- Enable GPI
- GPI Input A: EAS off/on
- GPI M (Metadata): Upmix L/Rf Off/On
- 2Ch = GPI M ON
- MD Fail = GPI M Off
- Primary Asmxot: 3/2L
- Current Metadata

PgmCfg Automation
- GPI High
  - Pgmcfg: 5.1+2
  - 1 – TV 5B Gen
- GPI Low
  - Pgmcfg: 2+2+2+2
  - 1 – <No Preset>
- GPI Setup

Nielsen Watermark *
- Encoder 1
  - Enable **v**
  - Audio Output: 1-6
  - SID: 09001
  - Check Digits: 0A
  - Distro: Final
  - Ins Mode: Override
- Encoder 2
  - Enable **v**
  - Audio Output: 1-6
  - SID: 09001
  - Check Digits: 0A
  - Distro: Final
  - Ins Mode: Override

Record Audio to USB
- Record **v**
- Channel mode: 8ch
- Inputs/Outputs/Both
- Duration: 1 hour

Test Tones
- Level: -20.0 dBFS
- Off, Sine, Voice, BLITS…
- Cycle Through 1-6

I/O Options
- PGM Config 5.1+2+Loc ***
- Downmix -> 7/8 ***
- Xtra Dly 1-2: 0-100ms
- Xtra Dly 7-8: 0-100ms
- Gain 7-8: 0 to -31dB
- Audio Inputs In 1/2 = AES In 1/2 In 9/10 = SDI In 9/10
- Dolby E Bypass **v**
- Comp Delay PCM **v**
- Enc B out = Enc B
- SDI Embed
- Clock Ref: SDI (SDI, AES1, Int 48k)
- SDI Outputs Out 1/2 = AES In 1/2 Out 15/16 = SDI In 9/10
- SIMM Source: AES1
- SIMM Byp. Src: AES1

Test Tones ->
- EAS Destination: PGM ***
- Clock Ref: SDI (SDI, AES1, Int 48k)
- Metadata: FAIL
- MD Source: RS-485
- VANC DID: 0x45
- VANC SDID: 0x01
- VANC Extract Line: 9
- VANC MD Async
- Lock Headphones
- Headphone Output Source: PGM1 (1-8) Downmix: LoRo (LtRt)

*Present in AERO.air (5.1) Version ONLY

Chapter 5: Presets & Processing Adjustment

The Linear Acoustic AERO.air 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.

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. For example, only presets labeled as "NR" will use the multiband source noise reduction feature.

Factory Presets

- **TV 5B GEN** - **T**ele**V**ision **5**-**B**and **GEN**eral. 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 LAMBDA or a Dolby LM100.

- **TV 5B LIGHT**- **T**ele**V**ision **5**-**B**and **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** - **T**ele**V**ision **5**-**B**and **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 LOUD** - **T**ele**V**ision **5**-**B**and **LOUD**. Similar to TV 5B HVY, but louder and more punchy.

- **FILM GEN**- Light processing appropriate for dialogue heavy films. Protection limiting acts when necessary, but audio is appropriately open.

- **SPORTS 5B GEN** - Medium processing with quick response and minimal background pumping. Keeps dialogue present and audible even in rich surround mixes.

- **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.

- **ITU 5B LOUD**ness **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.

- **MUSIC 5B HVY -** Aggressive music preset for music television type channels. Works well for delivering music programming to small televisions or portable receivers.

- **FINE ARTS 5B GEN-** Light processing for artistic programming that is already controlled in loudness.

- **Reference Settings -** Processing test settings

- **Reference Heavy -** Aggressive processing test settings

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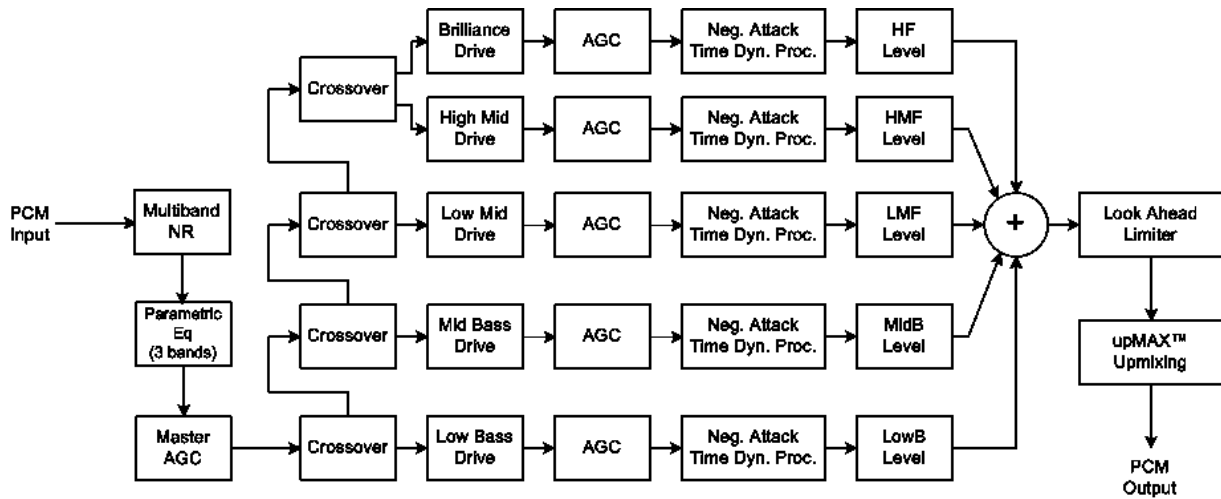
Figure 5-1 General signal flow of the AERO.air processing core

5.1.2   Adjusting Processing

Following is a description of all adjustable core processing parameters.
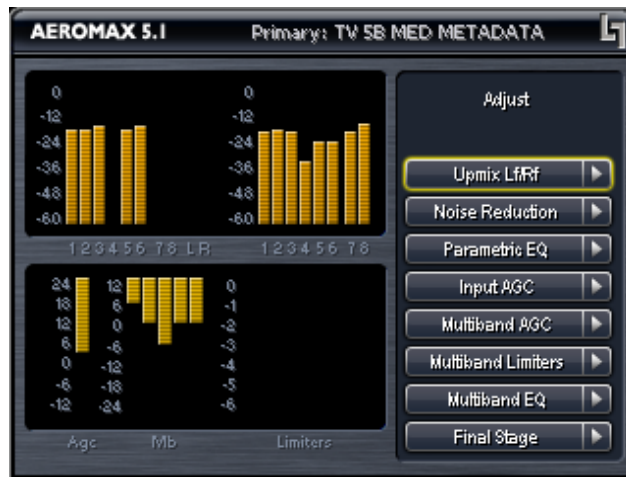
Figure 5-2 Adjust Processing Menu

But first... A Word on Our Crossovers

The multiband crossovers in the AERO.air 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 is 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 (Sets On-Air Loudness)

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 Loudness**: -36dB - 0dB

- A final output attenuator to calibrate loudness 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 that since this is the final attenuator, post all processing, this control also directly sets the maximum peak output of the unit, so a setting of -6dB means that maximum output will be -6dBFS.

NOTE: *Output Loudness should be adjusted to match the facility target or the dialnorm value of the Dolby Digital (AC-3) encoder.*

• **Current Output**: 0dB LKFS to -144dB LKFS

- A simple utility ITU-R BS.1770 loudness meter with 10 second integration; see below for detailed description.

5.1.3 Upmixing

As described in Chapter 4, upmixing is the process whereby two channel audio can be convincingly re-mapped into 5.1 channels using sophisticated level and phase detection techniques. The proprietary, industry standard and road-proven UPMAX® algorithm is incorporated in the AERO.air, and the new UPMAX® II algorithm can be added for enhanced spaciousness and resilience to low bitrate source material. The UPMAX family enables 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) from any encoding system. 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

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.

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:** Controls level 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 -10dB.

- **Use Center Width for 5.1 (On/Off):** Allows discrete center channel audio to also be spread across the front channels by the percentage set by Center Width. Useful for better integrating material mixed with only dialogue in the center channel. Cleverly conceived by our friends David and Darren at Ascent/Discovery UK, this mode is useful for content that has been produced with only dialogue in the center channel (allows for alternate languages). Default is Off.

- **Center Level:** (Active with UPMAX II option only) Allows minor adjustment of up-mixed Center channel level.

- **Center Exclusivity:** (Active with UPMAX-II option 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.

- **Algorithm:** (Active with UPMAX II option only) Selects between industry standard Linear Acoustic UPMAX and our newest version UPMAX II, both of which produce very consistent and downmix compatible results.

5.2   Calibrating Loudness (NEW)

Measuring loudness takes special meters and something that is often overlooked: it takes some time. While perception of loudness in humans seems nearly instantaneous, like the perception of color, it is subjective. To create a meter that produces readings that most listeners agree with has taken decades and has resulted in a metering standard that gives only a good estimation. But it is close enough for now, and good enough for success.

So, how do you set the output loudness of your AERO.air? Use a meter and a bit of common sense. We have taken care of the meter part: each processing core now contains a simple ITU-R BS.1770 meter with 10 second integration. The common sense part is up to you, and is where the idea of time comes in.

The goal of loudness metering and control is not to produce a single reading that does not move. That is valid only for test tones. For normal broadcast audio the goal is to produce a reading that is correct _on average_. With normal program audio applied to the AERO.air, navigate to the correct processing core (Modify Primary, Secondary, etc...), then go to Adjust/Final Stage and locate the Master Loudness control which will likely be set for -11dB. Now comes the common sense: watch the Current Output reading and LISTEN to the audio to make sure it is average program content: not screaming, not silence, but typical, average audio- normally a section with dialogue. If the meter shows your target loudness on average, you are done. If it does not, make small (i.e. 1dB) adjustments to Master Loudness control, then wait a few seconds for the meter to integrate the measurement, and repeat this until it the meter reads correctly.

NOTE: *For compliance, the loudness target should equal the dialnorm setting of the Dolby Digital (AC-3) encoder which may be internal or located downstream.*

We have been asked why we do not simply take this meter reading and feed it back to the Master Loudness control. The reason is simple: because this will un-do all of the previous careful processing that avoids pumping, breathing, and loudness wandering. The number will be correct, but the audio will not sound very pleasing.

5.3   Metadata

Metadata in the context of television audio is data that describes the audio data. Parameters describe program-specific things like the number of channels and the dialog loudness of a program. These values are intended to be set by the program producer, flow through the entire broadcast chain, and finally used to present a consistent audio experience to the consumer.

As more program producers begin to embrace some of the benefits of metadata, the system as a whole continues to improve, however protection against bad or missing metadata must still be provided. All it takes is for one errant program to have the wrong values or be missing values completely and the presentation suffers. Complaints of loudness shifts between commercials and programs, and image shifts between stereo and surround sound programs are still all too common.

The goal of Linear Acoustic products has always been to provide the best possible delivery of the original audio to consumers within the bounds of what is acceptable. Certainly a program that is produced very loudly and signalled as being very soft is pure abuse of the system and this must be prevented or at least minimized.

5.3.1   Metadata Structure

Audio metadata, also known as Dolby or Dolby compatible metadata can be broken down into two parts: professional and consumer. Professional audio metadata describes things like program configuration (i.e. the 5.1+2 seen on Dolby E or Linear Acoustic e-squared systems) and carried up to eight consumer metadata programs. The consumer metadata programs directly correlate to the program configuration, so 5.1+2 is two programs and each has its own set of consumer metadata parameters.

Consumer metadata contains the dialog loudness and audio coding mode parameters that can be used to preserve original audio quality. By using these parameters directly prior to Dolby Digital (AC-3) encoding, the consumer is presented with the best possible representation of the original audio, while not allowing loudness or image shifts.

For a more in-depth description of individual metadata parameters, please refer to the included ATSC document: IS-318 and visit:

http://www.linearacoustic.com/pdf/Metadata_Enc_Dec.pdf

5.3.2   Metadata Setup

The AERO.air 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 pro-

cessing and control upmixing respectively. Setup of metadata can be accomplished in the Modify Primary (or Modify Secondary) menu, under the Metadata sub-menu.

The setup procedure is described below:

- **Source: PGM1 to PGM8** - Describes which consumer metadata program will be used to control the processing. For an incoming 5.1+2 metadata stream, the Primary processing core should be set to respond to PGM1, while the Secondary processing core should be set to respond to PGM2 (if desired). Factory defaults are PGM1 for Main processing, and PGM2 for Secondary processing.

- **Apply Dialnorm: Yes/N**o - If set to yes, then the incoming dialnorm parameter is applied to the incoming audio to scale it prior to processing. Basically, 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.air will realign it in real time.

- **Global def dialnorm: 0 to 31** - Global default dialnorm determines what value should be used in case incoming metadata fails or is intentionally missing. Factory default is 24.

- **Metadata (status): Fail/OK** - indication of incoming metadata stream

    - **acmod:** 3/2L, 2/0, etc... per incoming metadata. Shows N/A in case of failure.

    - **dialnorm:** -1 to -31 per incoming metadata. Shows N/A in case of failure.

It is important when the system is set up that metadata be applied to the AERO.air only, and does not continue on to the Dolby Digital (AC-3) encoder, else the values will be doubly applied. Please see Chapter 3 for encoder setup information.

NOTE: The AERO.air should be the last device in the metadata chain to prevent double application of metadata values.

Chapter 6: Troubleshooting

The Linear Acoustic AERO.air 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.air 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.

6.1.1   Unit won't power on

First, check to see if the power supply fan is turning when AC power is applied. If the fan is spinning, remember that the boot process may take up to two minutes, and during the process you will hear the bypass relays being cycled. If the fan spins but the unit still does not boot all the way, switch it off, REMOVE THE AC CORD (a bias supply is still active even with the power switch off), wait two minutes, plug it back in and turn the unit on by pressing the Left, Up, and Right arrow keys.

6.1.2   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).

6.1.3   Dialogue Makes Surrounds Pump

It is possible that the chosen preset is too aggressive, too fast, and that Gate and Freeze are not set appropriately for the content. Note that all of the processing channels in a given program, such as 5.1, are linked and all get the same gain reduction and expansion. The loudest channel leads the gain control, so the perception is that the surrounds get ducked by the dialogue. We strongly suggest choosing an appropriate preset that is not too heavy

and then carefully adjusting Gate and Freeze to make sure that crowd noise is not increased when dialogue stops.

6.1.4   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.air is feeding is not properly referenced. Using the master bypass function of the unit is a helpful way to troubleshoot this issue.

6.1.5   Received Audio Has Dropouts

The primary cause of this is the Dolby Digital AC-3 encoder not being properly referenced with respect to the DTV 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 (i.e. set the encoder to "REF IN/SRC On mode). Consult your DTV encoder reference manual for more details.

6.1.6   Problems with EAS decoding

It is important to locate EAS encoding equipment after any audio processing, including AERO.air. It is critical that processing be minimized for use with current EAS systems. The AERO.air provides a special "post processing" insert specifically for EAS audio and it should be used.

6.1.7   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.air 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."

6.1.8   Audio Issues Disappear In Bypass

This can be caused by a number of things, including the fact that a processor like AERO.air 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!

6.1.9   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.

6.1.10   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 7: Specifications

Table 7-1 Electrical Specifications

Sampling Rate	48 kHz (±0.1%)
Output Delay	Baseband PCM outputs (1-10): 33 msec minimum; UPMAX II adds 20 msec and Nielsen Watermark adds 29 msec. Additional delay from Dolby encoding/decoding options, please see below.
Program Configurations	**AERO.air (5.1):** 5.1+2, 5.1+Local, 5.1+2+Local, 5.1+SAP, 2+2+2+2, **AERO.air (DTV):** 5.1 + Downmix EAS available in all configurations
Audio Word Length	Matches the AC-3 audio encoding system (24-bit capable); Internal processing is 32-bit.
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
Headphone Output	
Frequency Response	20 Hz–20 kHz, ±0.5 dB in Protection Limit
Distortion	Less than 0.2%, 20 Hz–20 kHz
Level	+12dBu Maximum
Output Connector	1/4" (6.35mm) front panel jack
Digital I/O	
Digital Audio Inputs	Seven unbalanced female BNC connectors, comply with AES-3ID-2001/SMPTE 276M. Internal 75-Ohm termination.
Digital Audio Outputs	Seven unbalanced female BNC connectors that comply with AES-3ID-2001/SMPTE 276M specifications.
REF Input	Unbalanced female BNC connector, complies with AES-3ID-2001/SMPTE 276M. Internal removable 75-Ohm termination.
SDI I/O	
HD/SD-SDI I/O	Up to 16 channels of audio, signals per SMPTE 292/299M-2004, VANC metadata 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 Audio Decoding Specifications

Audio Decoder Types	Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) per ATSC A/52:2010, Dolby E
Decoder Modes	Dolby Digital/Plus: up to 5.1 channels Dolby E: up to 8 channels plus metadata
Decoder Resolution	Up to 24-bit
Encoder Latency	Dolby Digital/Plus: 32 msec Dolby E: 32 msec NTSC, 40 msec PAL

Table 7-4 Mechanical Specifications

Dimensions	3.5 × 19 × 17 in (89 × 483 × 432 mm)
Net Weight	20 lbs (9 kg) approx.
Shipping Weight	25 lb (11.3 kg) approx.
Power Requirements	100-240 VAC, auto-sensing, 50–60 Hz
Power Consumption	100 W maximum

Table 7-5 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/Regulatory	North America: Designed to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Designed for U.S. and Canadian listing with UL. Europe: Designed to comply with the requirements of Low Voltage Directive 73/23/EEC and EMC Directive 89/336/EEC. Designed for RoHS and WEEE compliance.

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 7-6 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 7-7 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		