LA-5300 Broadcast Audio Processor

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Congratulations on your purchase of the Linear Acoustic LA-5300 Broadcast Audio Processor.

The LA-5300 provides everything broadcasters need to be ready for NEXTGEN TV/ATSC 3.0 audio in a single, compact, integrated package, including:

- Dolby® AC-4 encoding from PCM
- Transcoding from Dolby Digital and Dolby Digital Plus to Dolby AC-4
- Dolby AC-4 decoding for audience measurement watermarking and bitstream analysis and monitoring
- Linear Acoustic UPMAX® ISC upmixing
- Optional Verance and Nielsen watermarking
- Loudness control via Dolby Real-Time Loudness Leveler (when encoding to Dolby AC-4)

Optionally, it also provides processing for a second legacy ATSC 1.0 program, including:

- Dolby® Digital Plus (E-AC-3) and Dolby Digital (AC-3) encoding
- Linear Acoustic UPMAX® ISC upmixing
- Optional Verance and Nielsen watermarking
- Loudness control via Linear Acoustic APTO® or Dolby Real-Time Loudness Leveler

Quick Start Guide

Hardware Overview

LA-5300 Front Panel

Figure 1 - Front panel

The front panel of the LA-5300 includes the following:

- **Reset button** for restarting the unit from the front panel (1A)
- **Four status LEDs** indicating the **status of each power supply** (1B, 1C), the **overall status** of the unit (1D), and **sync for the reference clock** (1E)
- A **color LCD display** to show status and basic configuration parameters (1F)
- A **five-button navigation cluster** with Left, Right, Up, and Down arrows plus a green “OK” button (1G)
- A **USB port** used for software updates (1H)

The front panel is used for the initial setup and configuration including setting the IP address of the Control Ethernet port. Additional configuration is accomplished via the remote user interface.

LA-5300 Rear Panel

Figure 2- Rear panel

The rear panel includes the following connections:

- Two **RJ-45 Gigabit Ethernet connections**, one for network remote control (2A) and one for AES67 I/O (2B)
- Two independent **3Gb/s HD/SD-SDI inputs and outputs** (2C) on female BNC connectors
- **Quad-Link SDI** or **MADI I/O** (2D) - optional
- Five AES-3 I/O connections (2E) on female BNC connectors
- Parallel **GPI/O control port** (female DB-15) for five inputs and five outputs (2F)

- Two **IEC power inlets** for the redundant internal universal auto-ranging power supplies (2G)

> **Note -** Although the hardware is present for two independent SDI I/O paths, only the first SDI I/O is active at this time. The second path will be enabled in a future firmware update.

Installation and Initial Setup

Installation

LA-5300 is a 1RU product intended to be permanently installed in a standard 19½” equipment rack and secured with four standard rack screws. LA-5300 is fan cooled with air intakes and exhausts located on the side of the unit just behind the front panel, but whenever possible, it is recommended to leave 1RU of empty space above and below the unit.

Power

Plug the supplied IEC power cords into the LA-5300 and connect each to different mains power sources. Remember that while redundant supplies do protect against the unit losing power in the event of a PSU failure, the loss of mains supply voltage is a bigger concern. Accordingly, each supply should be fed from a different circuit equipped with adequate surge protection and fitted with an uninterruptible power supply (UPS).

> **Important -** Please click here for important information on proper grounding and other items pertaining to electrical safety.

Boot Up and Home Screen

The unit will power up as soon as the power cords are connected, and using the Reset button (A) is not required on initial startup. The front panel of the unit will remain dark during most of the boot process but the Status LED (D) will light. Once the boot process is complete, the Home Screen will appear and the Status LED will turn green.

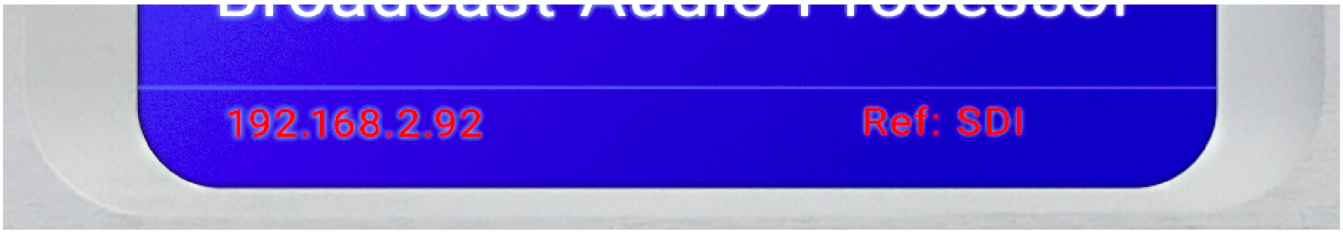

Figure 3- Front panel home screen

Setting IP Addresses

LA-5300 ships with DHCP enabled and will automatically retrieve an IP address when connected to a network with a DHCP server.

> **Important -** The Control and AES67 Ethernet connections both require 1000BASE-T (Gigabit) switch ports in order to work properly.
>
> If both the Control and AES67 ports will be used, they must be in different subnets. In-band control is available using the AES67 port.
>
> If only one port will be used, set the unused port to “Static IP” with an IP Address of 0.0.0.0, a Subnet Mask of 0.0.0.1, and a Gateway of 0.0.0.0.

To set a fixed IP address for the Control port:

- Press the **Right button** once to show the Information screen and once more to navigate to the IP Configuration (Control) screen
- Press the **Down button** to highlight “Edit”
- Press the **OK button** to highlight “Use DHCP”
- Press the **Right button** to highlight “Yes”
- Press the **Up/Down buttons** to select “No”
- Press the **Left button** to highlight “DHCP” again
- Press the **Down button** to highlight “Address”
- Press the **Right button** to move to the first digit of the IP address
- Use the **Up/Down buttons** to change the value
- Use the **Right/Left buttons** to select another digit to edit
- When finished editing the Address field, press the **Right/Left buttons** to highlight “Address” again
- Press the **Down button** to navigate to the “Netmask” field, following the instructions above for entering the value
- When finished editing the Netmask field, press the **Right/Left buttons** to highlight “Netmask” again
- Press the **Down button** to navigate to the “Gateway” field, following the instructions above for entering the value
- When finished editing the Gateway field, press the **Right/Left buttons** to highlight “Gateway” again
- Press the **Down button** to highlight “Update”

- Verify the information has been entered correctly and press the **OK button** to save the new values
- If you notice incorrect information and need to start over, use the **Right button** to highlight “Cancel” followed by the **OK button**

Figure 4 - System information screen

Figure 5 - IP configuration (control) screen

If you are using AES67 I/O, pressing the Right button from the IP Configuration (Control) screen will bring you to the IP Configuration (AES67) screen. The steps for configuring this port are identical to those above for the Control port.

Connecting Inputs and Outputs

The LA-5300 offers 3Gb/s HD/SD-SDI inputs and outputs, Quad-Link SDI or MADI I/O (optional and mutually exclusive), five AES-3 inputs and outputs on female BNC connectors, plus AES67 I/O via the

AES67 Ethernet port. Connect the physical inputs and outputs as appropriate for your installation.

Connecting to the Web-based User Interface

No special client software is required to remotely connect to the LA-5300, and the HTML-5-based GUI is device, operating system, and browser agnostic. We recommend Google Chrome or Apple Safari. The computer or tablet should be on the same network and subnet as the LA-5300.

Once both LA-5300 and your computer are connected to your network, enter the IP address of the LA-5300 in the URL field of your browser. Once connected, the **Home screen** will appear

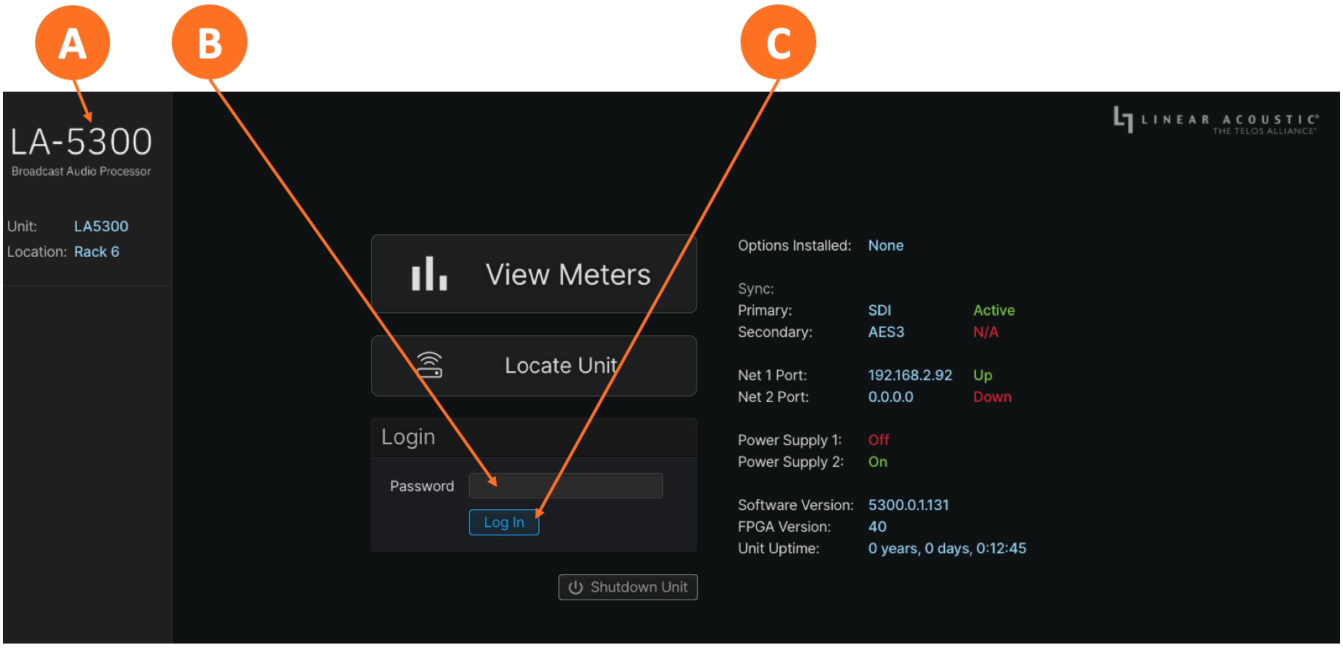

Figure 6- Home screen

Clicking on the **LA-5300 logo** (6A) will always bring you back to the Home screen. To log in to the unit, enter the default password of **1234** into the **Password field** (6B) then click the **Log In button** (6C). The ability to change the password will be included in an upcoming software update.

> **Note -** The Linear Acoustic LA-5300 is manufactured under license from Dolby Laboratories. Dolby, Dolby Audio, and the double-D symbol are trademarks of Dolby Laboratories.

User Manual

Introduction

The LA-5300 provides everything broadcasters need to be ready for NEXTGEN TV/ATSC 3.0 audio in a single, compact, integrated package, including:

- Dolby® AC-4 encoding from PCM
- Transcoding from Dolby Digital and Dolby Digital Plus to Dolby AC-4
- Dolby AC-4 decoding for audience measurement watermarking and bitstream analysis and monitoring
- Linear Acoustic UPMAX® ISC upmixing
- Optional Verance and Nielsen watermarking
- Loudness control via Dolby Real-Time Loudness Leveler (when encoding to Dolby AC-4)

Optionally, it also provides processing for a second legacy ATSC 1.0 program, including:

- Dolby® Digital Plus (E-AC-3) and Dolby Digital (AC-3) encoding
- Linear Acoustic UPMAX® ISC upmixing
- Optional Verance and Nielsen watermarking
- Loudness control via Linear Acoustic APTO® or Dolby Real-Time Loudness Leveler

> **Note -** The LA-5300 is also available in a decode and monitor-only version which omits upmixing, watermarking, and AC-4 encoding (which includes loudness control). Accordingly, some menus, screens, and controls will look different from those seen here.

Front and Rear Panel Overview

LA-5300 Front Panel

Figure 1 - Front panel

The front panel of the LA-5300 includes the following:

- **Reset button** for restarting the unit from the front panel (1A)
- **Four status LEDs** indicating the **status of each power supply** (1B, 1C), the **overall status** of the unit (1D), and **sync for the reference clock** (1E)
- A **color LCD display** to show status and basic configuration parameters (1F)
- A **five-button navigation cluster** with Left, Right, Up, and Down arrows plus a green “OK” button (1G)
- A **USB port** used for software updates (1H)

The front panel is used for the initial setup and configuration including setting the IP address of the Control Ethernet port. Additional configuration is accomplished via the remote user interface.

LA-5300 Rear Panel

Figure 2- Rear panel

The rear panel includes the following connections:

- Two **RJ-45 Gigabit Ethernet connections**, one for network remote control (2A) and one for AES67 I/O (2B)
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> **Note -** Although the hardware is present for two independent SDI I/O paths, only the first SDI I/O is active at this time.

Installation and Initial Setup

Installation

The LA-5300 is a 1RU product intended to be permanently installed in a standard 19½" equipment rack and secured with four standard rack screws.

LA-5300 is fan cooled with air intakes and exhausts located on the side of the unit just behind the front panel, but whenever possible it is recommended to leave 1RU of empty space above and below the unit.

Power

The LA-5300 has two internal auto-ranging redundant power supplies protected by type T1AL25V fuses. Remember that while redundant supplies do protect against the unit losing power in the event of a PSU failure, the loss of mains supply voltage is a bigger concern. Accordingly, each supply should be fed from a different circuit equipped with adequate surge protection and fitted with an uninterruptable power supply (UPS). Providing the receptacles to which the supplied power cords are connected are grounded, the power connection serves as a ground. Additionally, the rear panel earthing terminal should be connected to a proper ground system with 20 – 14 AWG (0.5 – 1.5mm) wire.

> ⚠ **Important -** Please click here for important information on proper grounding and other items pertaining to electrical safety.

Boot Up and Home Screen

The unit will power up as soon as the power cords are connected, and using the Reset button (A) is not required on initial startup. The front panel of the unit will remain dark during most of the boot process but the Status LED (D) will light. Once the boot process is complete, the Home Screen will appear and the Status LED will turn green.

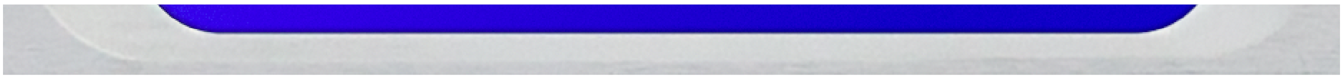

Figure 1- Front panel home screen

Setting IP Addresses

LA-5300 ships with DHCP enabled and will automatically retrieve an IP address when connected to a network with a DHCP server.

> **Important -** The Control and AES67 Ethernet connections both require 1000BASE-T (Gigabit) switch ports in order to work properly.
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- When finished editing the Gateway field, press the **Right/Left buttons** to highlight “Gateway” again
- Press the **Down button** to highlight “Update”
- Verify the information has been entered correctly and press the **OK button** to save the new values
- If you notice incorrect information and need to start over, use the **Right button** to highlight “Cancel” followed by the **OK button**

Figure 2 - System information screen

Figure 3 - IP configuration (control) screen

If you are using AES67 I/O, pressing the Right button from the IP Configuration (Control) screen will bring you to the IP Configuration (AES67) screen. The steps for configuring this port are identical to those above for the Control port.

Locate Mode

Our sincere hope is that you like your LA-5300 so much that you'll eventually end up with dozens of them populating your racks, but since one LA-5300 looks pretty much like another, we've come up with a way to help you identify an individual unit.

When you enable the “Locate Mode” from the remote user interface, the unit’s front panel display will repeatedly change color until Locate Mode has been cleared by pressing any key on the front panel or by clicking again on the “Locate Mode” button on the remote user interface. Enabling Locate Mode affects only the unit’s display and has no effect on the audio.

Factory Reset

To reset the unit to its factory defaults, simultaneously press and hold the Left, Up, and Right buttons for ten seconds until a red countdown timer appears in the bottom left corner of the display.

> **Important!** Performing a factory reset will erase all I/O routing as well as any other custom settings you may have saved. This information cannot be reclaimed once you reset the LA-5300, even by contacting customer support or feeling genuine remorse for your decision. We strongly recommend backing up your configuration to a computer before performing a factory reset to avoid re-setter’s remorse. You have been warned!

Hardware Relay Bypass

LA-5300 includes a hard relay bypass that removes all internal circuits and processes from the signal path and connects each SDI and AES-3 input directly to its corresponding output.

To toggle the relay bypass on and off, simultaneously press and hold the **Left**, **Down**, and **Right keys** for three seconds until a red countdown timer appears in the bottom left corner of the display. **Do not confuse this sequence with the Factory Reset procedure described above.**

Remote User Interface

Connecting to LA-5300

As outlined in the section on Installation and Initial Setup, certain settings and configurations can be accessed and performed using the front panel buttons and screen, but the LA-5300 is designed to be connected to a network and operated through its web-based interface.

> **Important!** As explained in the section on Setting IP Addresses, LA-5300 must be connected to 1000BASE-T (Gigabit) switch ports, and the Control and AES67 ports should not be connected to a common network with the same netmask/subnet.

No special client software is required, and the HTML-5-based GUI is device, operating system, and browser agnostic. We recommend using Google Chrome or Apple Safari. The computer or tablet should be on the same network and subnet as the LA-5300.

> **Note -** This section is primarily an overview of the user interface to provide an explanation of the various menus and screens. It is not intended to provide step-by-step instructions for setting up or operating the LA-5300. Where applicable, references to specific chapters will be made where you can find more detailed instructions for specific controls.

Once the LA-5300 and your computer are connected to your network, enter the IP address of the LA-5300 to which you want to connect in the URL field of your browser. Once connected, the Home Screen will appear. As you navigate through the user interface, clicking on the **LA-5300 logo** (1A) in the top left corner will always return you to this screen.

Multiple users can connect to the same LA-5300 hardware simultaneously. Anyone connected to the unit will be able to see the information on this screen including the **unit's name** (1B) and **physical location** (1C).

Additionally, all users can access the **Meters screen** (1D) which shows input and output level meters along with pertinent information about the input or output bitstream (depending on the workflow selected). See the section “LA 5300 Home Screen When Logged In” below for detailed information.

The **Locate Unit feature** (1E), which causes the front panel display of the unit to flash for easy identification in a crowded rack, can also be activated without logging in.

Additional information about installed options, clock sync, network status and IP address, power supply status, software and FPGA versions, and unit uptime **are displayed to the right of the screen** (1F).

User Accounts and Logging In

Making modifications to the LA-5300's configuration, signal routing, or transcoding parameters via the remote user interface requires logging into the unit. At this time, only one level of access is supported.

Enter the default password (1234) in the **password field** (1G) and click on the **Log in** button (1H). The password can be changed on the System page.

To remotely power down the unit, click on the **Shutdown Unit button** (1I).

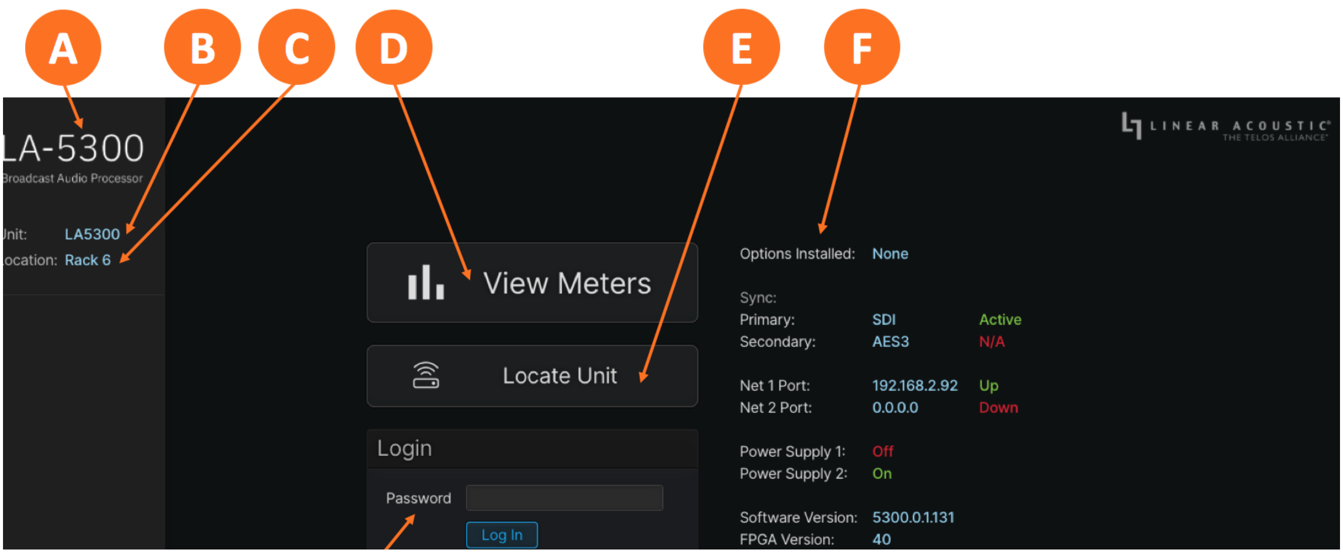


Figure 1 - Home screen before logging in

Additional menus become visible once you are logged in. These include the encoding, upmixing, watermarking, and monitor controls for the **Program 1 (Dolby AC-4) encoder** (2A), the same controls for the **Program 2 (Dolby Digital Plus) encoder** (2B), the **I/O menus** (2C) which include clock, GPI/O, and delay controls, **detailed system-wide information and configuration settings** (2D) which includes the IP configuration menu, and the **Log Out button** (2E).

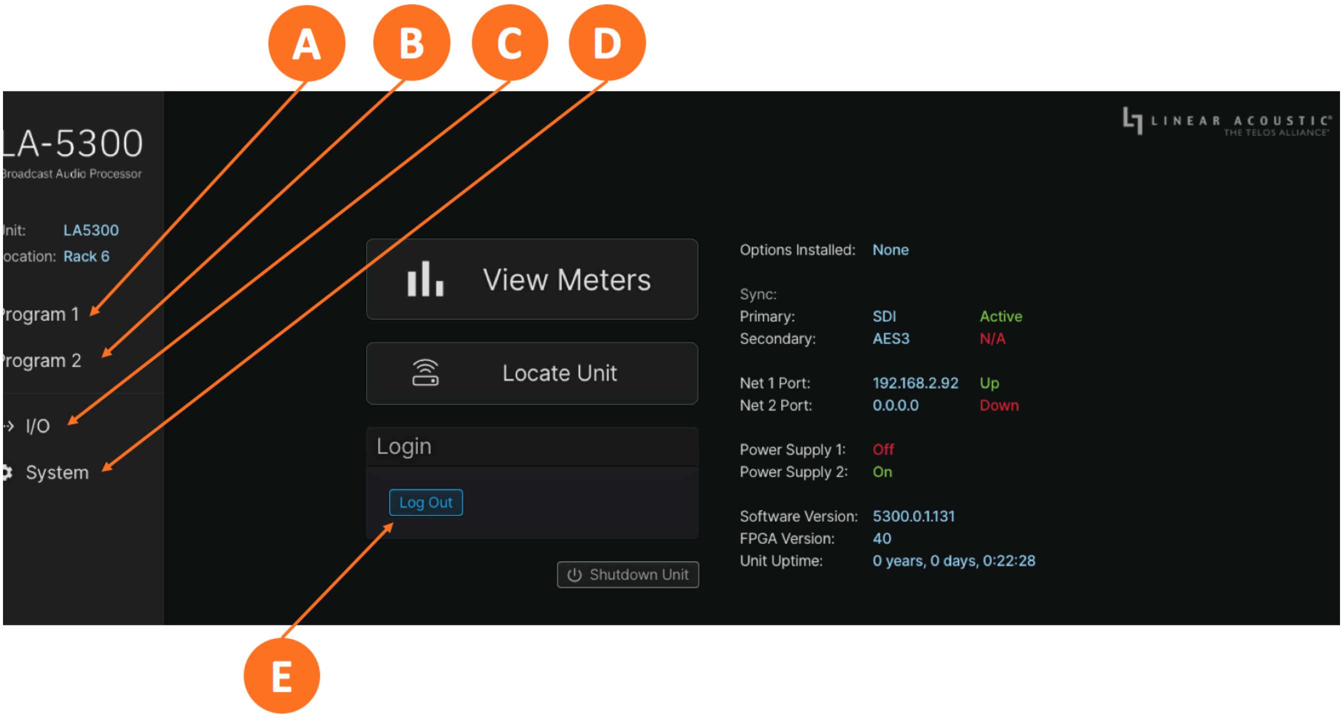


Figure 2 - Home screen when logged in

Program 1 and Program 2 Menus

Clicking on the **Program 1 menu** (3A) brings up the main menu for all signal processing and monitoring functions related to Program 1. The name of the program can be customized by clicking on the **program text field** (3B).

There is a dropdown menu to expose the individual controls for **Dolby AC-4 encoding** (3K), **upmixing** (3J), **Nielsen watermarking** (3I), **Verance watermarking** (3H), and **confidence monitoring** (3G).

Detailed operation of each process is outlined in subsequent sections.

Specific settings for all Program 1-related processing - Dolby AC-4 encoding, upmixing, Nielsen watermarking, Verance watermarking, and confidence monitoring - can be saved in a single program preset in the **Preset menu** (3C) for easy recall. Note that there are no factory presets, and so the menu field will appear blank until a user preset is created and saved.

Clicking on the **Preset management icon** (3E) reveals a dropdown menu with the following options:

- **Save:** Saves the current Program 1 settings to the selected preset, immediately over-writing the previous settings associated with the preset; note that if there are no user presets saved, the "Save As" dialogue box will appear and prompt you to enter the name of the new preset
- **Save As:** Saves the current Program 1 settings to a new preset; a "Save As" dialogue box will appear and prompt you to enter the name of the new preset
- **Delete:** Permanently deletes the selected preset
- **Import:** Allows you to import a preset that was previously exported and saved to the remote PC; this can be a handy tool when setting up multiple units that are configured in exactly the same way rather than setting each control individually
- **Export:** Allows you to export a preset either for safekeeping as a backup should it accidentally get erased from the unit, or for making it available for uploaded to other units requiring the same configuration using the "Import" feature described above

This page also includes **input meters** (3E) and **output meters** (3L) for each channel of each input type – SDI, AES67, and AES-3. The meters for each input type can be minimized or expanded.

Clicking on the **Program 2 menu** (3F) reveals identical controls for the second program which includes Dolby Digital Plus encoding for creating an E-AC-3 or AC-3-encoded bitstream for legacy ATSC 1.0 support.

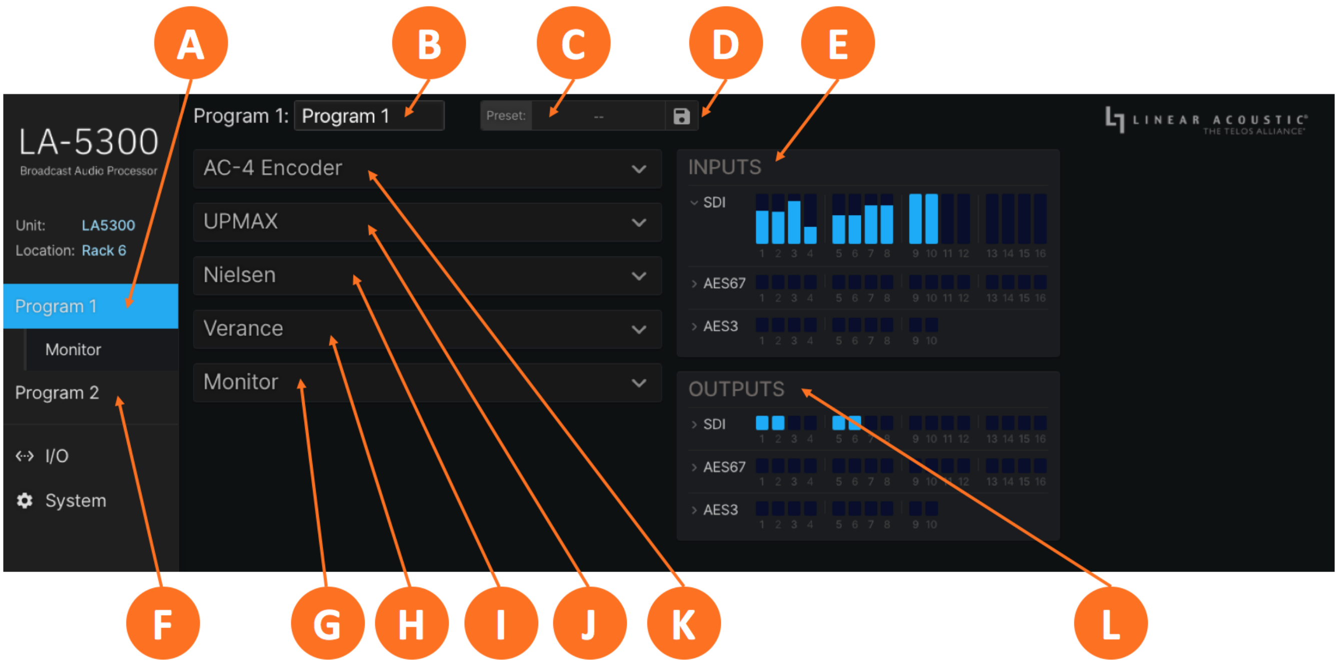


Figure 3- Program menu

I/O, Clocking, and Reference

Input and Output Routing

Connect your SDI, AES-3, and/or AES67 sources to the rear panel of the LA-5300 as required for your installation.

The LA-5300 can access any of the eight audio pairs carried in the incoming HD- or SD-SDI streams applied to SDI Input 1. Inputs and outputs for five stereo pairs via AES-3 are also provided.

Any visible and accessible AES67 networked audio source can be input into LA-5300 for processing then returned to the network via AES67, embedded into the SDI output, and/or sent to the AES-3 output. LA-5300 supports taking an input signal from one source type and sending the resulting audio or bitstream to another output type.

Initial I/O Setup

Log into the LA-5300 and click on the **I/O menu** (1A). Choose between SDI/AES-3/MADI or AES67 inputs and outputs in the **Workflow menu** (1B). Select between viewing Program (processed) and Passthrough (unprocessed) routing with the **Program/Passthrough switch** (1C).

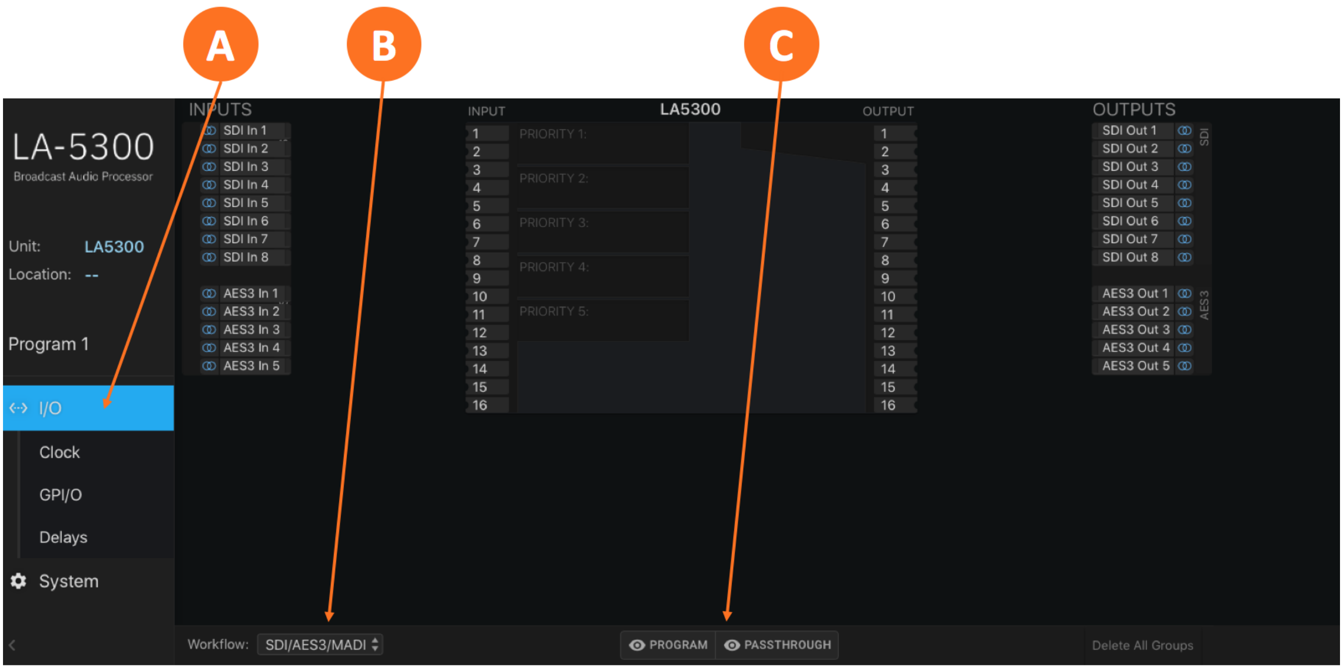


Figure 1 - I/O menu

Routing a signal into the LA-5300 is a two-step process.

First, create an input group from the **available physical hardware inputs** (2A). Next, **route the audio from these inputs into the processor input** (2B).

Routing a signal to the output is a similar two-step process.

First, create an output group from the **available physical hardware outputs** (2D). Next, r**oute the audio from processor output** (2C) to the physical outputs.

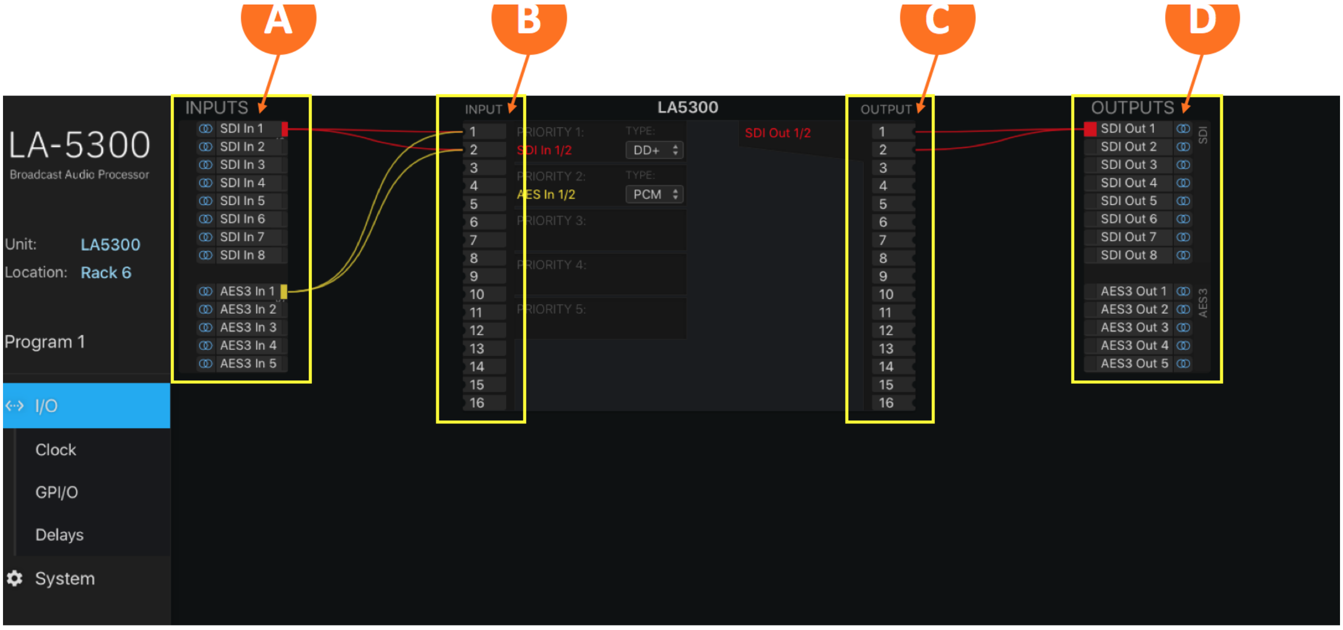


Figure 2 - Signal routing menu

Creating an Input Group

Click on your first **audio input source** (3A) to highlight it, then right-click and select “Create Input Group.” You may enter a custom name for this source in the **Label field** (3C) and **customize the color** (3B) of the graphical routing lines.

Multi-channel groups can be created by holding the “Shift” key while selecting sources (Shift + Click).

Creating groups using non-contiguous channels is accomplished by holding the “Ctl” key while selecting sources (Ctl + Shift).

> **Note -** The channel order must still follow the proper convention to be properly assigned. For example, L/R, C/LFE, Ls/Rs, Lfh/Rfh, Lrs/Rrs if setting up a 5.1-channel input on SDI 1-6 with height channels on SDI 13-16.

Click the **Save button** (3D) to save the configuration or the **Cancel button** (3E) if you need to start over without saving your configuration.

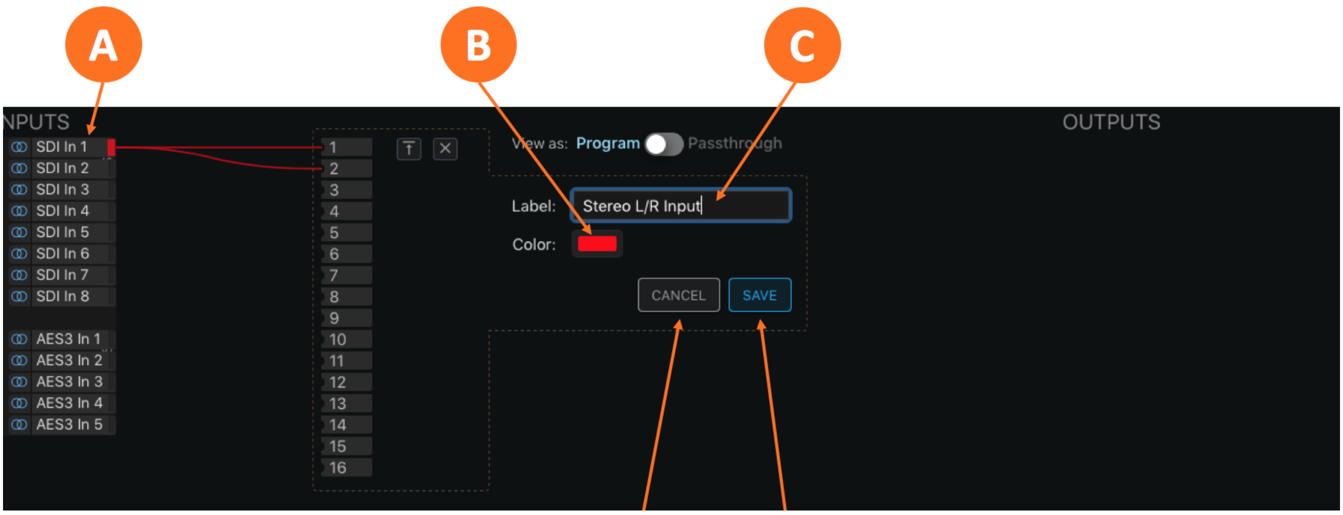


Figure 3- Creating an input group

Routing Audio to the Program Input

After saving the input group, it is necessary to first set the priority of the input source by dragging the **input audio lines** (4A) and then clicking when the **desired priority level** (4B) is highlighted. The LA-5300 will look for audio first on inputs connected to Priority 1. If no signal is present, it will look to subsequent priorities in order.

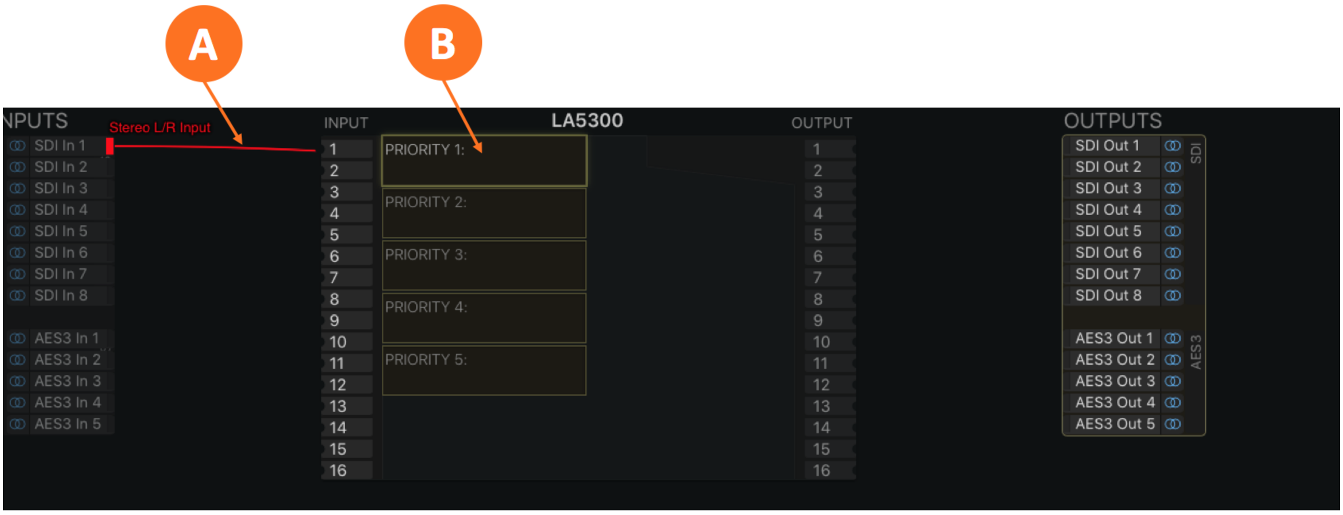


Figure 4 - Routing input audio and setting the priority

Next, choose the input signal type (PCM, DD+, EAS PCM or AC-4) from the **Type dropdown menu (**5A).

> **Note -** Whenever EAS PCM is used, it should be set as Priority 1 to ensure EAS alerts always have priority over any other program.

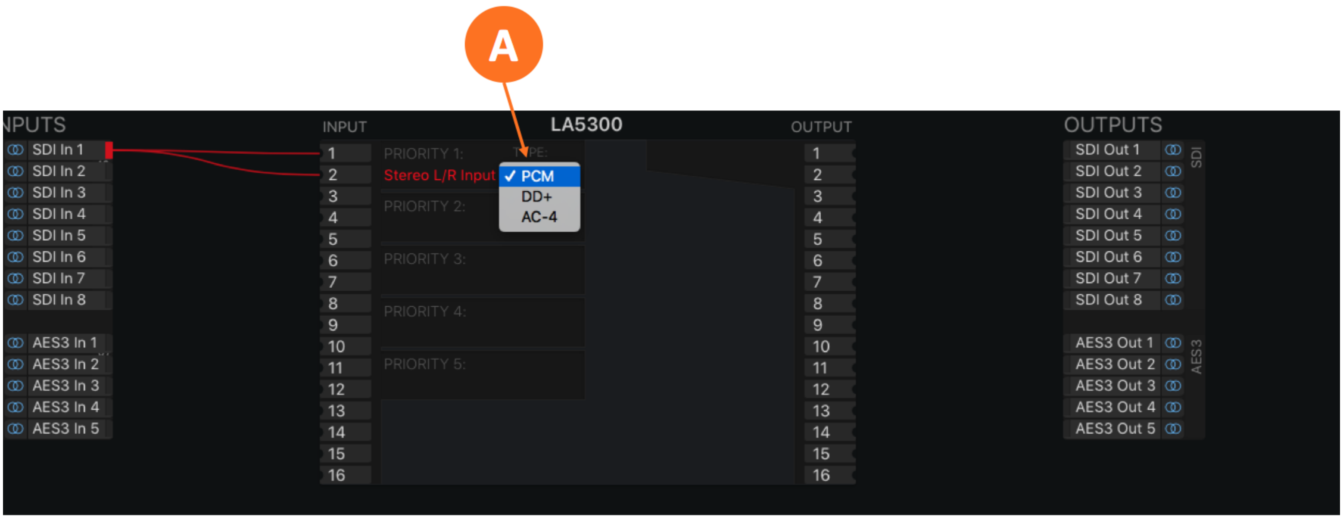


Figure 5 - Selecting the input signal type

To edit or delete a group, highlight then right-click the **colored Edit rectangle** (6A).

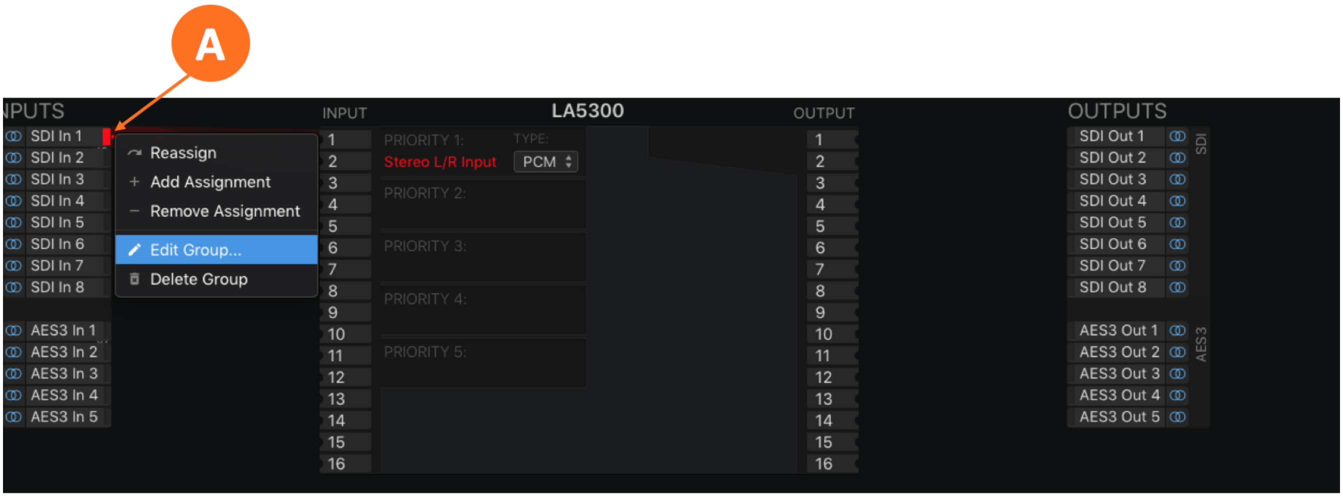


Figure 6 - Editing or deleting an input group

Creating an Output Group

The steps for creating an output group are identical to creating an input group, but with the Outputs menu.

Routing Audio to the Program Output

Once the first output group has been created and saved, drag the output audio lines from the group to the desired program output, then click to save.

Latency and Delays

Latency through the LA-5300 varies depending on which inputs and outputs are used and whether or not SRCs are enabled. In any case, it will be necessary to compensate for any latency incurring in the encoding process. Please see the Specifications section for latency measurements for the current software version.

To compensate for the processor's latency and avoid lip-sync issues, an SDI video delay is provided in the **Delays sub-menu** of the I/O menu (7A). Click on the millisecond and/or microsecond fields in the **Video Delay section** (7B) and type in the required value. If the output needs to be delayed by an even frame (the value of which will vary depending on frame rate), simply enter that number.

> **Note -** Entering an even frame value will result in a situation where it is necessary to add additional audio delay to the AES67 and AES-3 outputs so that they match the timing of the SDI output signal. This can be accomplished by setting the **Audio Delay** (7C) to a value equal to the difference between a full frame and the latency of the unit.

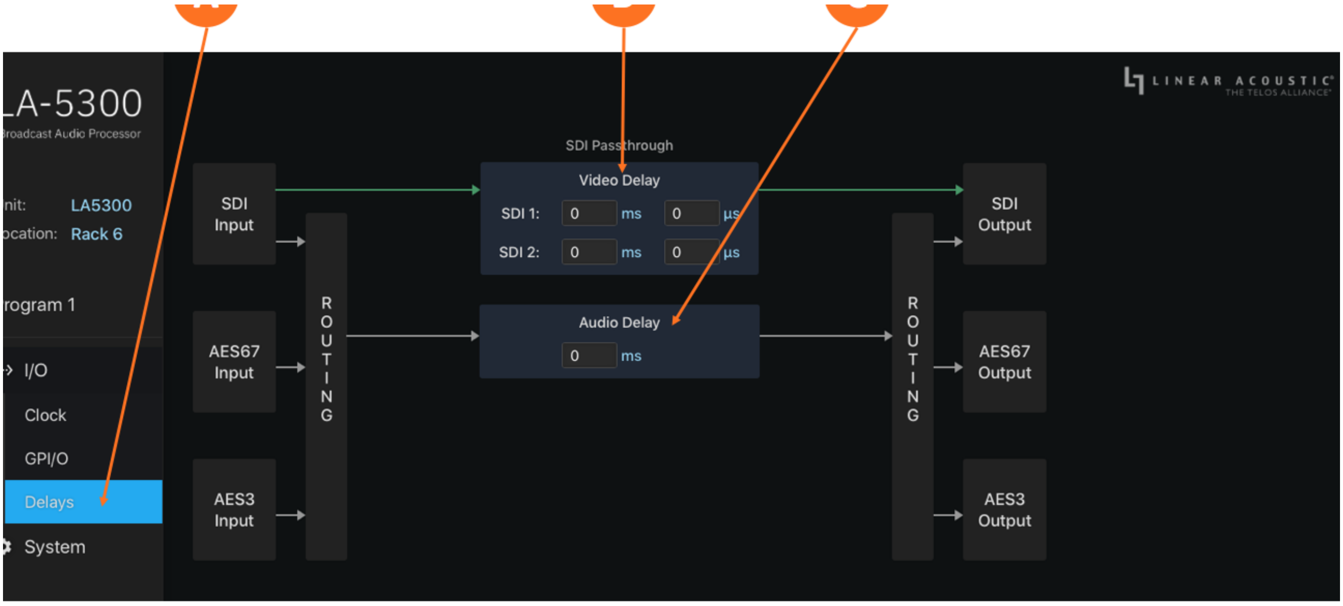


Figure 7 - Delays menu

Sample Rate Converters

Sample rate converters (SRCs) are provided on each SDI input pair, each SDI output pair of the first SDI output, and each AES-3 input pair. The AES-3 output is always synced to the active reference clock. **There are no SRCs in the AES67 path.**

The SRCs can be enabled and disabled per pair by first highlighting then clicking on the appropriate input or output pair, and then selecting **Channel Settings** (8A).

> **Important!** Only PCM audio can be passed through a Sample Rate Converter. Coded audio bitstreams including Dolby Digital Plus and Dolby AC-4 cannot be passed through an SRC without corrupting the data stream.

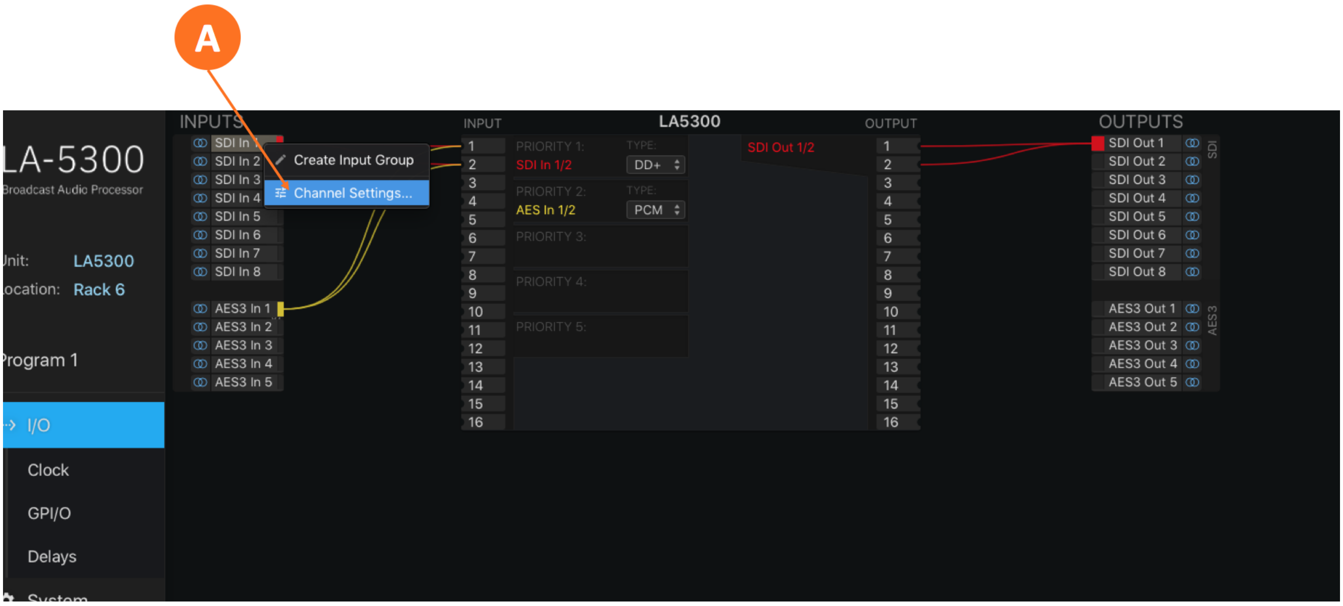


Figure 8 - Sample rate converters

Clock Reference

Because the LA-5300 offers very flexible signal routing and supports multiple input and output formats, having a firm grasp of the clock sync (reference) requirements is critical.

Clock Reference Requirements

Reference clock source options include:

- Internal 48kHz
- SDI Input
- AES-3 Input
- MADI Input (when optional MADI card is present)
- PTP (AES67)

When all input sources are SDI and the output of the processor is routed to the embedded SDI output, the reference signal present on the SDI input must be used.

When using AES-3 I/O, the LA-5300 can be referenced to either the AES-3 clock (using the SRCs in the SDI path) or to the SDI clock (using the SRCs in the AES-3 path).

When using an AES-3 source without an accompanying reference, the LA-5300's 48kHz internal clock can be used as the system reference if necessary, providing the SRC on the AES-3 input is enabled.

Whenever AES67 audio is used either on the input, the output, or both, the LA-5300 **must** be slaved to an externally-generated PTP clock as it cannot generate its own PTP reference.

> **Note -** The LA-5300 is set to operate in slave mode by default. There are controls in the Sync/ QoS sub-menu of the System menu to change this should the ability to operate as a clock master be introduced in future software versions, but only Slave Only mode is currently supported.

Setting Clock Reference

Click on the **Clock menu** (9A). Use the dropdown menus to choose a **Primary Reference Clock** and a **Secondary Reference Clock** (9C) and, if needed, a **VRef clock source** (9E).

The **status** of the selected Primary and Secondary Reference Clocks is shown to the right of the dropdown menu (9D). The absence or presence of a signal for each input type along with the format, resolution, and frame rate for incoming SDI signals is shown in the **Input Status section** (9B).

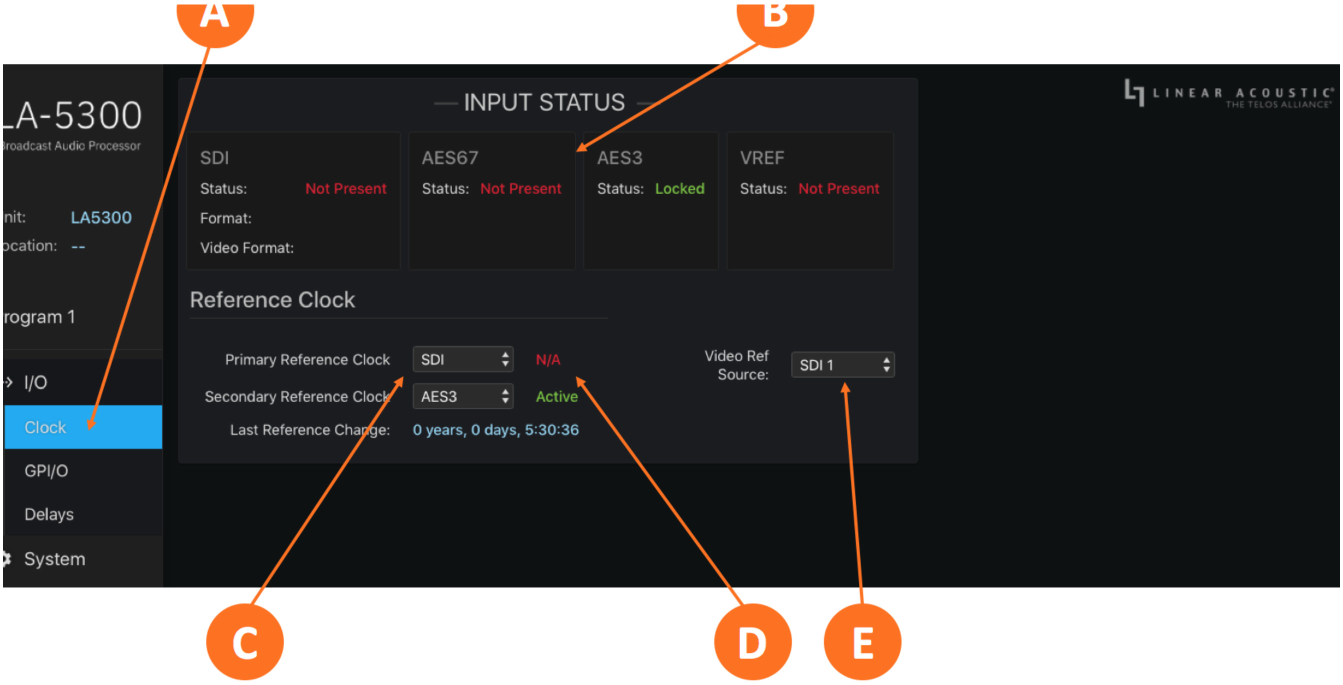


Figure 9 - Clock reference

GPI/O

LA-5300 offers five GPI and five GPO functions through its rear panel DB-15 connector, activated by a momentary contact closure.

GPI/O Pinout

Pin	Function	Pin	Function
Pin 1	+5VDC	Pin 9	+5VDC
Pin 2	GPI 1	Pin 10	GPO 1
Pin 3	GPI 2	Pin 11	GPO 2
Pin 4	GPI 3	Pin 12	GPO 3
Pin 5	GPI 4	Pin 13	GPO 4
Pin 6	GPI 5	Pin 14	GPO 5
Pin 7	Ground	Pin 15	Status OK
Pin 8	Ground		

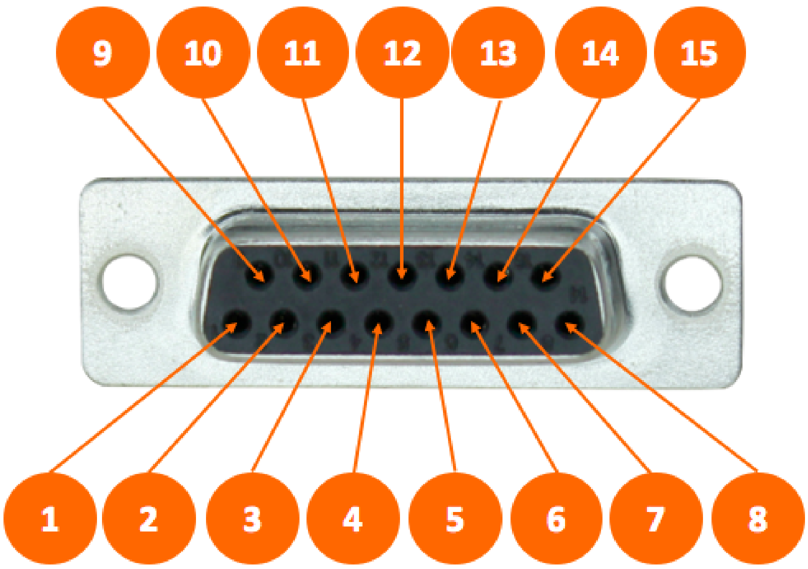


Figure 10 - GPI/O pinout

GPI/O Functions

GPI functions (11A) include:

- None
- Reboot/Reset Unit
- Hardware (Relay) Bypass
- Processing/Encoding Preset
- I/O Preset
- Show Preset (Processing/Encoding + I/O presets)

GPO functions (11D) include:

- None
- GPI Passthrough
- Power Supply 1 up
- Power Supply 2 up
- Power Supply 1 down
- Power Supply 2 down
- Unit resetting
- Unit rebooting
- Unit powering down

- Unit above temperature
- Unit bypass (relay bypass) active
- Change in reference
- Primary reference clock is lost
- Secondary reference is lost
- Primary reference clock is active
- Secondary reference is active
- Internal reference is active

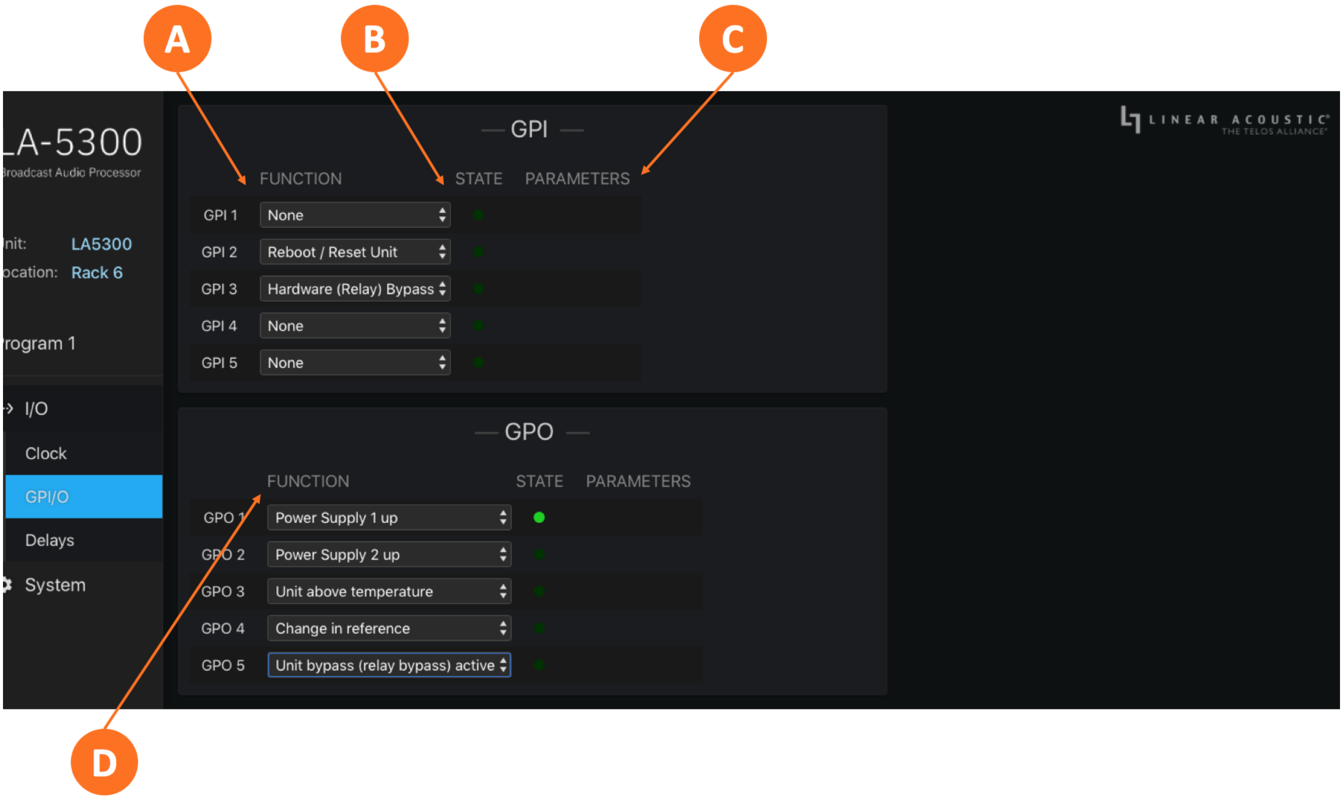


Figure 11 - GPI/O menu

The **State Indicator** (11B) for each function will light for the duration of the closure on GPIs. For GPOs, it will light for the duration of the active event.

At this time, there are no user-defined **Parameters** (11C) associated with any GPI or GPO. Should functions with such parameters be included in future software releases, they will appear in a dropdown menu.

System and Network Configuration

System Menu Overview

The **System Menu** is home to the System status screen. It also contains the menus and controls for configuring the Control and AES67 Ethernet ports, and setting up Sync and QoS when using a PTP clock.

System Screen

The main system screen contains a wealth of technical **status information about the device hardware and software** (1A). Some of it relates to the overall health of the unit (such as Power Supply status and unit uptime) but most information will only be required during the course of troubleshooting an issue with our support team or performing software updates.

The **Unit Name and Location** (1B) are entered on this screen. These are “friendly” fields to be used in any way that makes sense for your particular operation. Click inside each field to make it editable, enter the information, then click on the green checkmark to save.

Clicking on the **Master Unit Bypass button** (1C) engages a hard relay bypass that removes all internal circuits and processing from the signal path and connects each AES-3 and SDI input directly to its corresponding output. This button will turn red when the LA-5300 is in bypass mode. Note that this same hard relay bypass automatically engages when the unit is powered down.

Clicking on the **Locate Unit button** (1D) will cause the front panel of this particular LA-5300 to change color, making it easy to identify it amongst the dozens of them we hope will populate your racks. A second click will return the display to normal. Note that enabling Locate Mode will not affect the audio.

The **Reboot Unit control** (1E) reboots the unit. During the reboot process, the bypass relay will engage.

> **Important!** During the reboot process – which typically takes less the one minute – no processing will take place. Depending upon your I/O configuration, a complete loss of audio may occur.

Selecting the **Reset Unit to Factory Defaults button** (1F) will return the unit to its factory configuration, minus IP settings which are, by default, retained. To include IP settings in the reset, enable the **Include IP Settings in Reset control** (1G).

> **Important! Performing a factory reset will erase all I/O routing as well as any other custom settings you may have saved. This information cannot be reclaimed once you reset the LA-5300, even by contacting customer support or feeling genuine remorse for your decision.** We strongly recommend backing up your configuration to a computer before performing a factory reset to avoid re-setter’s remorse. You have been warned!

The LA-5300 has **two software banks** (1H). Both can be populated with different software versions, but only one version can be active at any given time. The active bank is indicated in red. Instructions for uploading and selecting different software versions are provided along with the software when you either download it from the customer portal or website or it is provided by our customer support team.

The factory default password is 1234 but can be changed in the **Set Password menu** (1I) by entering the current password then entering and confirming the new password.

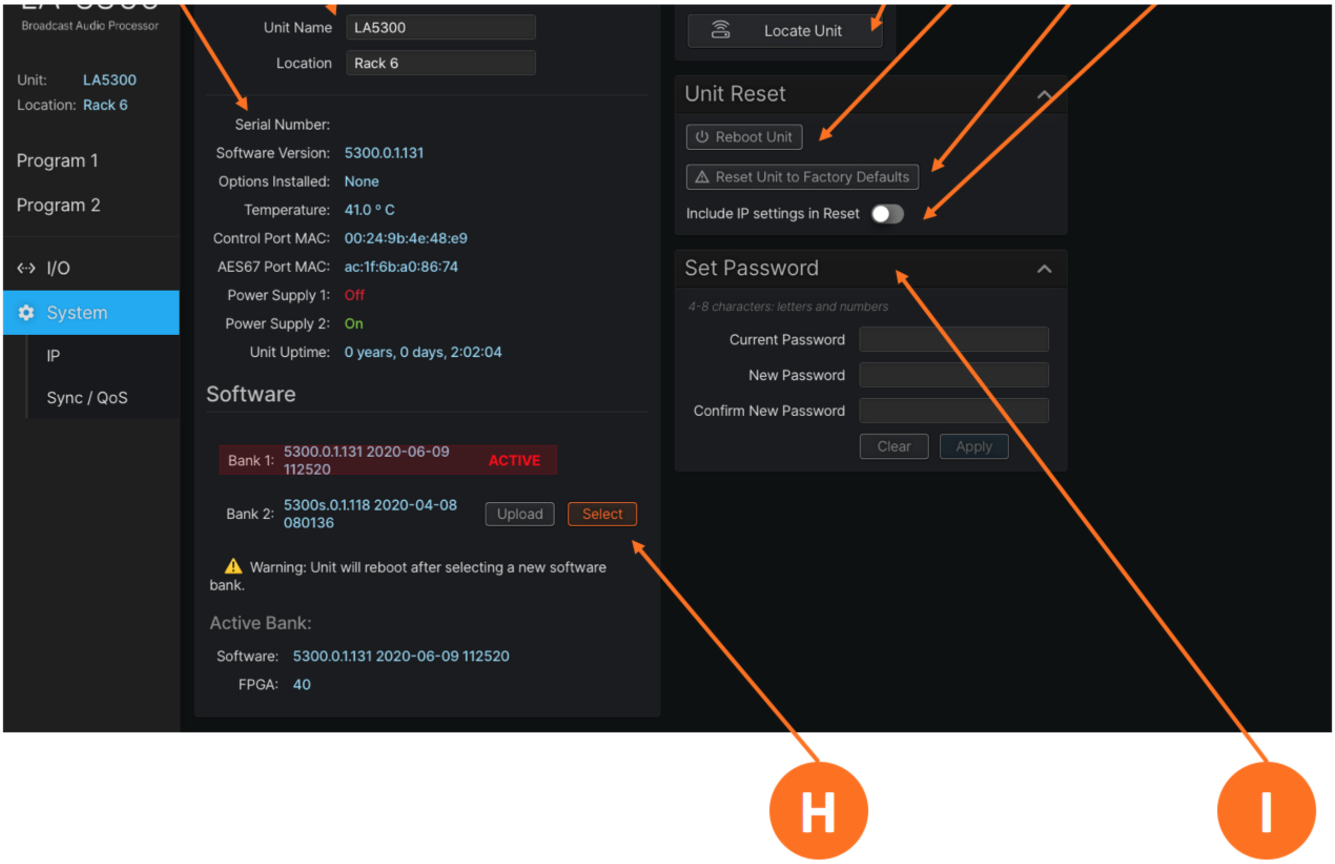
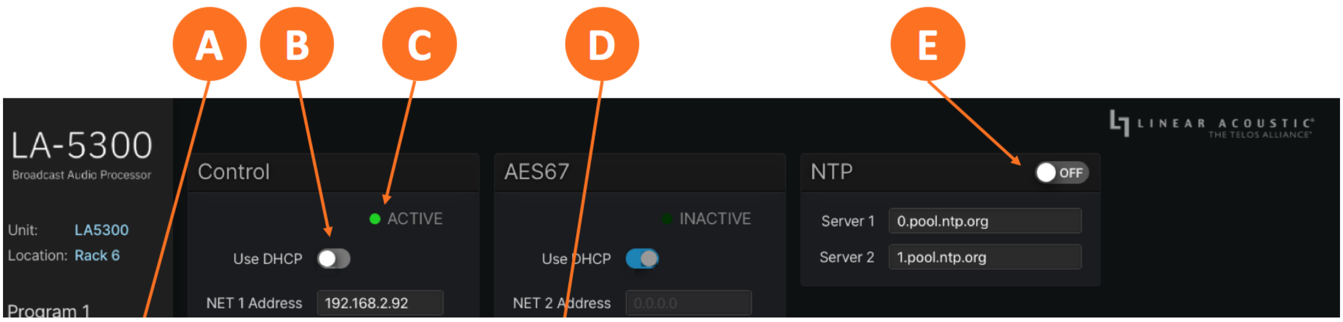


Figure 1 - System screen

IP Configuration

Clicking on the **IP Menu** (2A) reveals the controls for DHCP settings plus fields for entering static IP addresses, subnets, and gateways for both the Control and AES67 Ethernet ports as well as network activity and status. It also contains the control and server address information required by NTP.

> **Important!** The Control and AES67 Ethernet connections both require 1000BASE-T (Gigabit) switch ports in order to work properly.
>
> If both the Control and AES67 ports will be used, they must be on different subnets. In-band control is available using the AES67 port.
>
> If only one port will be used, set the unused port to “Static IP” with an IP Address of 0.0.0.0, a Subnet Mask of 0.0.0.1, and a Gateway of 0.0.0.0



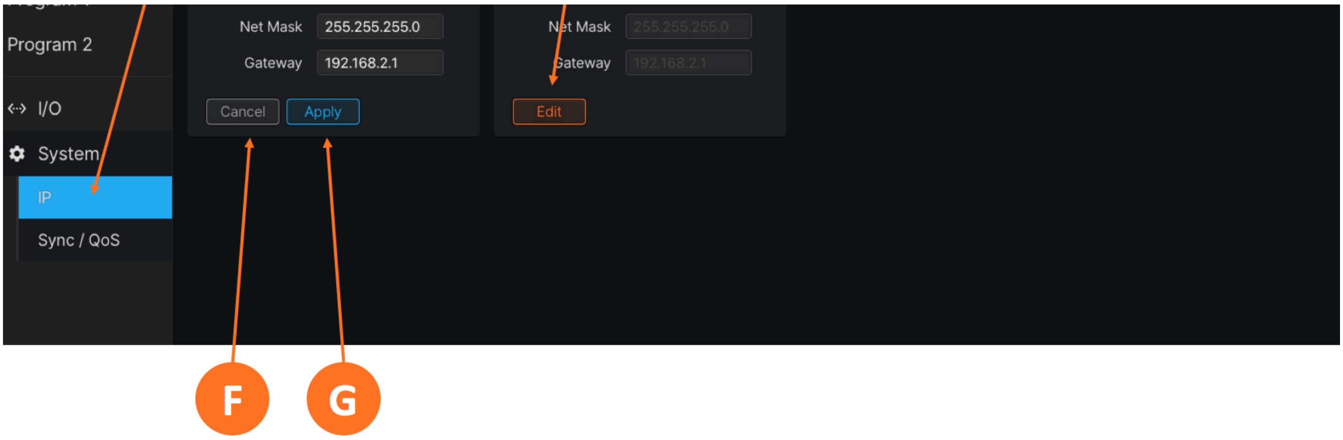


Figure 2 - IP configuration menu

By default, LA-5300 is set up with DHCP enabled and will automatically receive an IP address when its Control Ethernet port is connected to a network with a DHCP server.

To use a fixed IP address, click the **Edit button** (2D) for the appropriate port which unlocks the settings. (Please note the **Edit** button of the AES67 port is used for illustration in Figure 2 and is replaced by the **Cancel** and **Apply** buttons once you click on it).

Click on the **Use DHCP button** (2B) to disable DHCP. The text of the **NET Address**, **Net Mask**, and **Gateway fields,** which is grayed out when DHCP is enabled, will turn white and become editable. Once the information is entered, click on the **Apply button** (2H) to save the information or the **Cancel button** (2G) to back out without saving any changes. Verify that you have a good network connection by looking for a green dot and the word **Active** in the status window (2C).

Setup is identical for both the Control and AES67 ports.

To sync the LA-5300 to an NTP server, enable the **NTP control** (2E). By default, both the Server 1 and Server 2 fields are populated with ntp.org addresses. However, we recommend entering the IP addresses of local in-network NTP servers instead as the delay incurred in the process of reaching an internet server and then negotiating firewalls and other security measures when re-accessing your local network may be long enough to cause issues with accurate Nielsen and Verance watermark encoding measurements.

Dolby® Encoding and Monitoring

The LA-5300 provides standard Dolby AC-4 encoding in support of ATSC 3.0 for its primary program (Program 1).

Optionally, it can encode a second program (Program 2) to Dolby Digital Plus (E-AC-3) and Dolby Digital (AC-3) to provide support for ATSC 1.0 services.

Clicking on the Dolby AC-4 Encoder dropdown menu from the Program 1 screen reveals the basic encoder controls. There is a separate dropdown menu for advanced controls.

Clicking on Program 2 in units so equipped provides similar controls for the Dolby Digital Plus encoder.

Basic Dolby AC-4 Encoder Menu

Basic parameters include **Channel Mode** (1A), **Frame Rate** (1B), **Bit Rate** (1C), audio **Type** (1G), **Loudness Regulation** (1F), **Language** (1E), and **Dialogue Normalization** (1D).

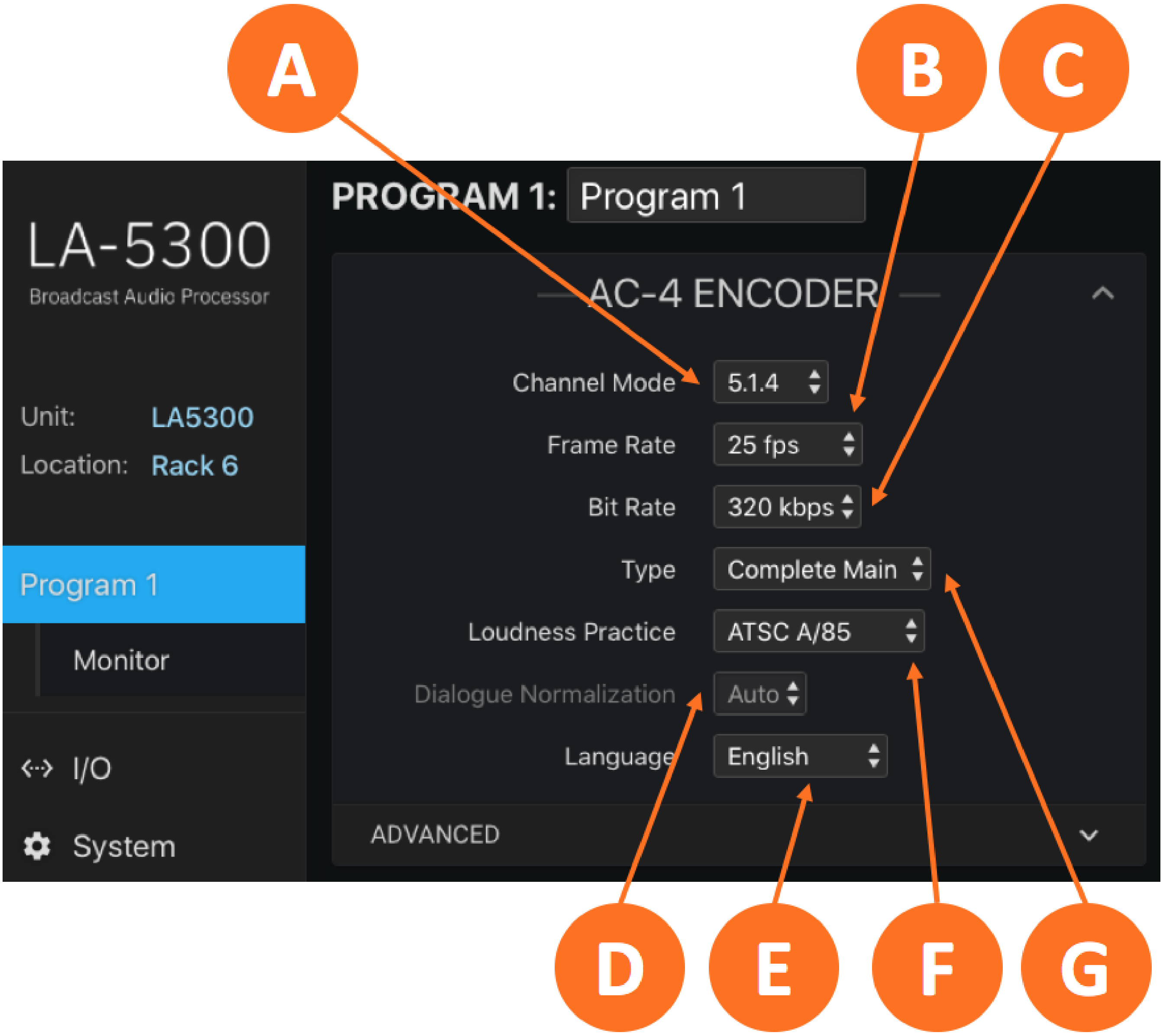


Figure 1 - Basic Dolby AC-4 encoder menu

Channel Mode

The **Channel Mode control** (1A) sets the configuration of the audio output channels embedded in the AC-4 bitstream. Choices include 2-channel (stereo), 5.1-channel, or 5.1.4-channel outputs (5.1-channel “base layer” with ear-level channels + 4 overhead channels).

Frame Rate

The Dolby AC-4-encoded audio frames must be aligned with video frames to prevent losing audio frames or introducing A/V sync errors during source switching, much like Dolby E or ED2. The **frame rate control** (1B) should be set to match the video frame rate in your particular workflow. Select “Native” when there is no video reference available.

Bit Rate

The Dolby AC-4 codec provides increased efficiency compared with Dolby AC-3 (Dolby Digital) to allow for the delivery of Dolby Atmos and multi-language content. The optimal bitrate for any given application – set by the **Bit Rate control** (1C) - will depend on a number of variables including channel mode and additional dialogue and music and effects content in NGA applications, but the basic recommendations are listed below.

Channel Configuration	Recommended Dolby AC-4 Data Rate	Equivalent Dolby AC-3 Data Rate
Stereo (2/0)	64 kbps	192 kbps
5.1	144 kbps	382 kbps
5.1.4	288 kbps	N/A

Type

AC-4 can deliver a traditional single mix, consisting of dialogue, music, and effects elements. This is referred to as a “complete mix” and often abbreviated as “CM”.

It can also deliver a single common music and effects element (“M&E”) together with separate dialogue elements (“D”) with each dialogue element selected and mixed with the M&E element in the AC-4 decoder. This ability is what allows for “personalized audio” as different elements can be combined into multiple audio presentations and delivered within a single bitstream.

Local emergency audio can also be delivered in the bitstream.

The **Type control** (1G) identifies whether the stream is Complete Main, Music and Effects, or Dialog. Currently, only Complete Main is supported.

> ⓘ **Note -** Emergency Audio is signaled by selecting EAS audio as the Priority 1 source when configuring input groups. Please see the section on Routing Audio to the Program Inputs for more information.

Loudness Practice

The AC-4 codec includes integrated loudness management via the Dolby Real Time Loudness Leveler. RTLL normalizes the incoming audio signal prior to encoding. Presets provided in the **Loudness Practice control** (1F) for the following regulations:

- **EBU R 128:** Used primarily in Europe, audio is normalized to a target of -23dB LUFS as calculated over the entire duration of a program (integrated loudness) and without regard for isolating and measuring dialogue; it specifies a deviation of +/- 0.5LU and a peak level below -1dBTP

- **ATSC A/85:** Used in the United States and serving as the foundation and reference for the CALM Act; whenever possible, the anchor element of the audio (typically dialogue) should be measured and normalized to a target of -24dB LKFS, +/- 2dB which is accomplished here by enabling the dialogue intelligence feature
- **ARIB TR-B32:** The Japanese broadcast standard that uses a relative gate (per ITU BS 1770-2, like EBU R 128) but with a target loudness level of -24 LKFS and a maximum True Peak of -1dB.
- **Free TV OP-59:** Used in Australia, this standard is based on the previous OP-48 regulations (which measured VU and digital peak levels and required a target level of -20dBFS) but instead measures average perceived loudness using ITU-R BS 1770-3 with a target of -24dB LKFS and a maximum True Peak of -2dB.

Setting the control to “Manual” allows the target loudness value to be manually set with the Dialogue Normalization control.

Dialogue Normalization

When using one of the pre-defined Loudness Practice profiles, the **Dialogue Normalization control** (1D) is set to “Auto” and grayed out, and dialogue normalization will be enabled or disabled per the selected profile. When the Loudness Practice is set to “Manual”, a specific loudness target can be set.

In addition, the Dialogue Intelligence control – described below in the “Advanced” menu section – can be turned on or off.

Language

While the AC-4 codec supports the delivery of content in multiple languages, the primary language must be identified using the **language dropdown menu** (1E).

Advanced AC-4 Encoder Menu

The advanced parameters menu contains controls that determine whether certain settings and metadata values are set automatically or manually, and, if set manually, the values of individual settings.

Please note the advanced menu scrolls within the web page and not all controls are visible at once. The screenshots and control descriptions below are divided into “top” and “bottom” figures for ease of illustration and explanation.

Preferred Downmix Method

The **Preferred Downmix Method control** (2A) sets the downmixing metadata value for the downstream decoder.

- **Auto:** Uses the upstream metadata to set the preferred downmix
- **Not Indicated:** No instructions to the downstream decoder are specified, allowing the decoder or the end user to choose the downmix type

- **Lo/Ro:** The Center and surround channel content is redirected to the Left and Right channels for playback in a system with only two speakers
- **PL:** This is a Dolby Surround-compatible matrix-encoded downmix that contains content from all channels and downmixes them to the Left and Right channels; this downmix is intended to be decoded by a Dolby Surround Pro Logic decoder which can extract the matrix-encoded content from the Left and Right channels to produce a four- or five-channel output with mono surround channels
- **PLII:** This is a Lt/Rt downmix that retains the stereo surround information from the original program; a Dolby Pro Logic II-compatible downmix is intended to be decoded by a Dolby Pro Logic II decoder.

Settings Mode

The **Settings Mode control** (2B) determines whether the **device-specific DRC profiles** (1C) and various **channel-specific downmix values** (2D) are automatically or manually set.

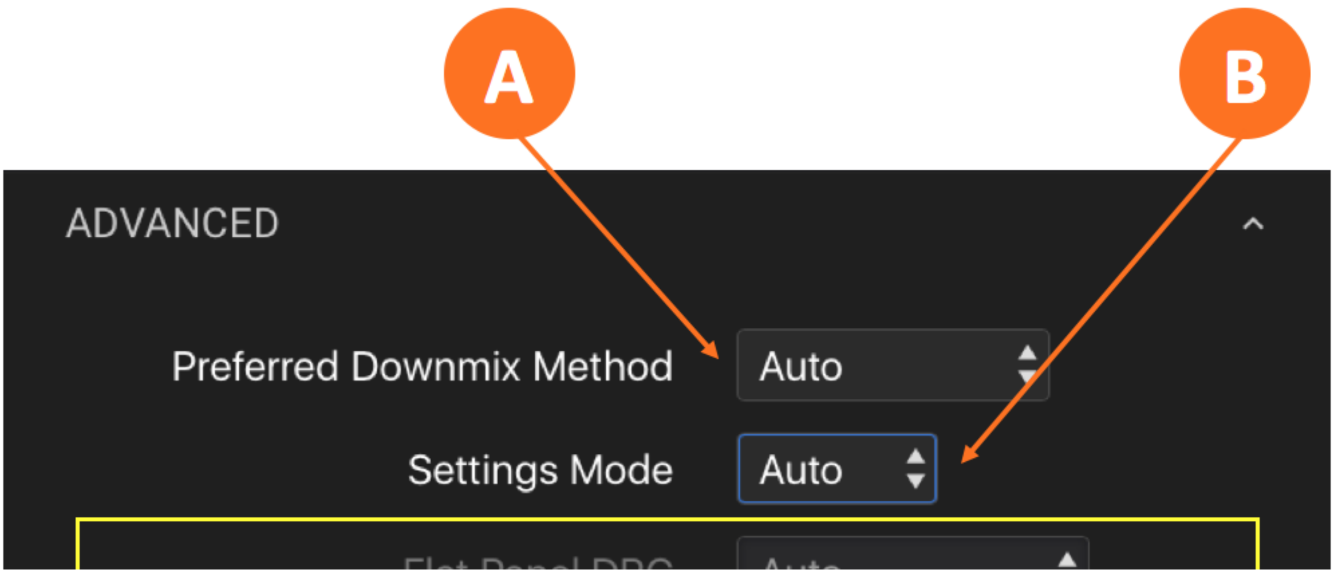
When set to Auto, controls for device-specific DRC profiles as well as downmix modes are grayed out.

Selecting "Manual" allows DRC profiles to be individually set for **Flat Panel televisions, Home Theater setups, Portable Headphones, and Portable Speakers** (2C).

DRC profiles include:

- Auto
- None
- Film Standard
- Film Light
- Music Standard
- Music Light
- Speech

Likewise, downmix gain values ranging from -6dB to +3dB can be individually specified Lo/Ro Surround, Lo/Ro Center, Lt/Rt Surround, and Lt/Rt Center Downmix configurations from a range of choices in each dropdown menu. Setting any of these controls to "Auto" will allow them to be set by upstream metadata.



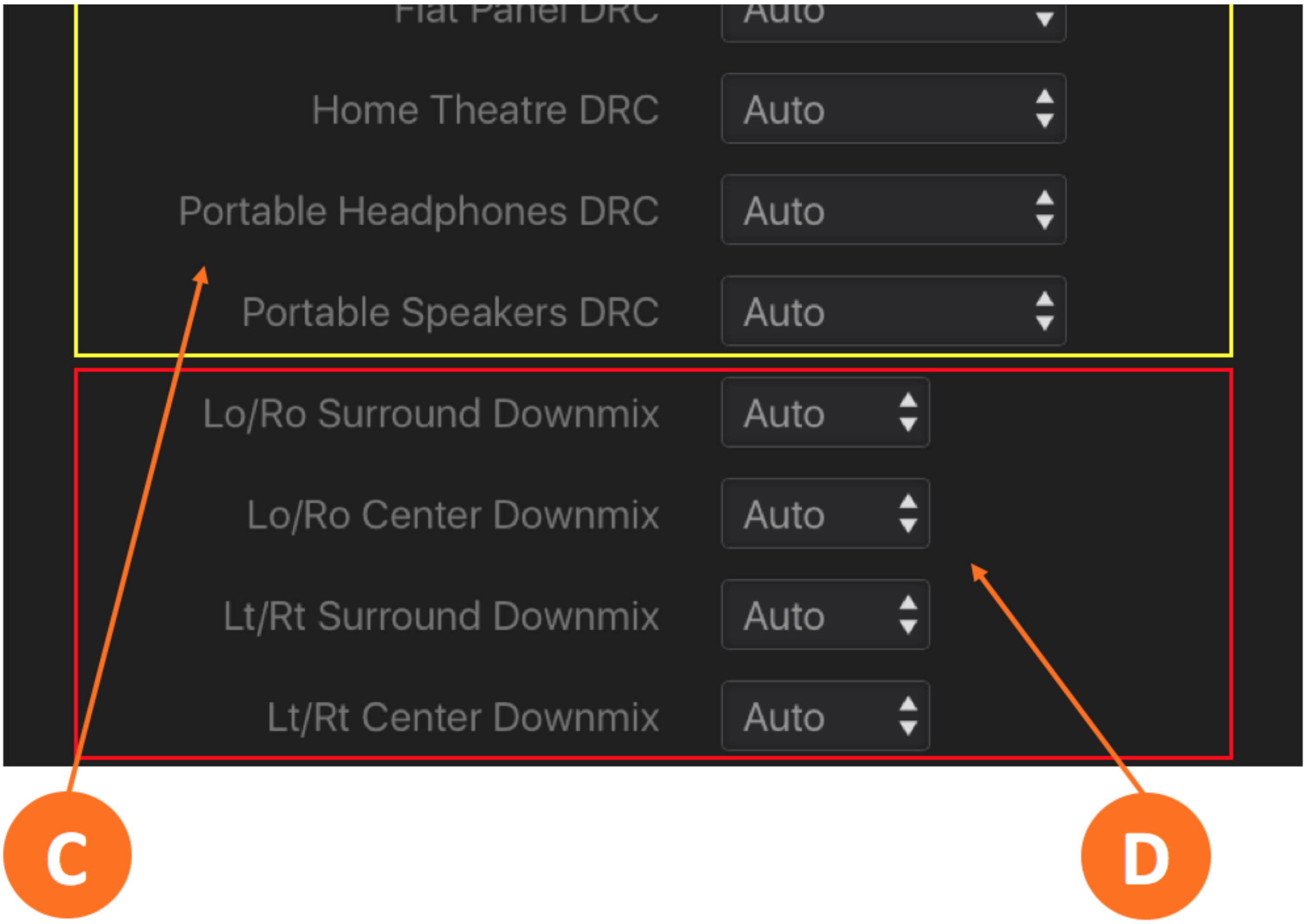


Figure 2 - Top portion of Dolby AC-4 encoder menu

Preprocessing Mode

Setting the **Preprocessing Mode control** (3A) to “Auto” grays out the associated controls in the “Previous Parameters” (3B) section and uses upstream metadata to determine individual control settings. Placing it in “Manual” allows each parameter to be manually and individually controlled.

Previous LFE Filter

“Applied” indicates LFE filtering has already been applied upstream and will not be added here. “Not Applied” indicates the filtering has not been applied upstream and will be added by the encoder. “Unknown” should be selected in situations where it is not known whether or not filtering has been applied.

Previous Phase 90 Filter

“Applied” indicates a 90-degree phase shift has already been applied to the surround channels upstream and will not be added here. “Not Applied” indicates the phase shift has not been applied upstream and will be added by the encoder. “Unknown” should be selected in situations where it is not known whether or not the phase shift has been applied.

Previous Surround Attenuation

“Applied” indicates surround attenuation has already been applied upstream and will not be added here.

“Not Applied” indicates surround attenuation has not been applied upstream and will be added by the encoder. “Unknown” should be selected in situations where it is not known whether or not surround attenuation has been applied.

Previous Mix Type (2 channel)

If the downmix type is known and present in the incoming metadata stream, set this control to “Auto” to pass the metadata through to the decoder. Choose “Unknown” in situations where the previous mix type is not known.

Choosing “Mix Down Lo/Ro”, “Mix Down PL”, or “Mix Down PLII” will force the decoder to use the specified mix type.

Previous Mix Type (5 channel)

5-channel content may have been initially downmixed from a higher channel count or upmixed from a lower channel count. If this information is known and present in the incoming metadata stream, set this control to “Auto” to pass the metadata through to the decoder. Choose “Unknown” if the previous mix type is not known.

Selecting “Mix Down PLIIx”, “Mix Down PLIIx Movie”, “Mix Down PLIIx Music”, or “Mix Down PLIIz” will force the decoder to use the specified mix type for content that has been mixed down from a high channel count.

Choosing “Mix Up PL”, “Mix Up PLII Movie”, “Mix Up PLII Music”, or “Mix Up PLII Professional” will force the decoder to use the specified mix type for content that has been upmixed from a lower channel count.

LFE Monitor Level

LFE signals are typically boosted by 10dB during the encoding process to ensure effects in the LFE channel are properly presented in the mix and deliver the desired impact. If this boost has already been applied upstream, select “Original Level” in the **LFE Monitor Level control** (3F). If it has not been applied upstream, select “Boost +10dB”. Selecting “Not Indicated” provides no specific information on this parameter to the downstream decoder.

Dialogue Intelligence

The setting of the **Dialogue Intelligence control** (3E) is automatically set to comply with the specific loudness regulation selected with the Loudness Practice control. However, it can always be manually enabled and disabled.

Note that as soon as a different Loudness Practice is selected, the Dialogue Intelligence control will reflect the settings the chosen profile.

Loudness Control Amount

When using one of the included Loudness Practice profiles, the amount of loudness control provided by the Dolby Real Time Loudness Leveler is automatically set to deliver a compliant output and the **Loudness Control Amount control** (3C) will be grayed out.

When the Loudness Practice control is set to “Manual”, the amount of processing performed by the RTLL can be manually set. Lower settings provide more gentle level control and are best suited for content that has already been processed for compliance either in the file domain or by a real-time processor upstream. Higher settings gradually increase the amount of processing and are more suitable when program content levels vary. A setting of “0” defeats the RTLL completely.

Loudness Limit Mode

Peak loudness values can be read on a sample-by-sample basis or with the use of over-sampling for a more granular measurement. Setting the **Loudness Limit Mode control** (3D) to “True Peak” will over-sample peak measurements in compliance with ITU-R BS.1770-4 recommendations.

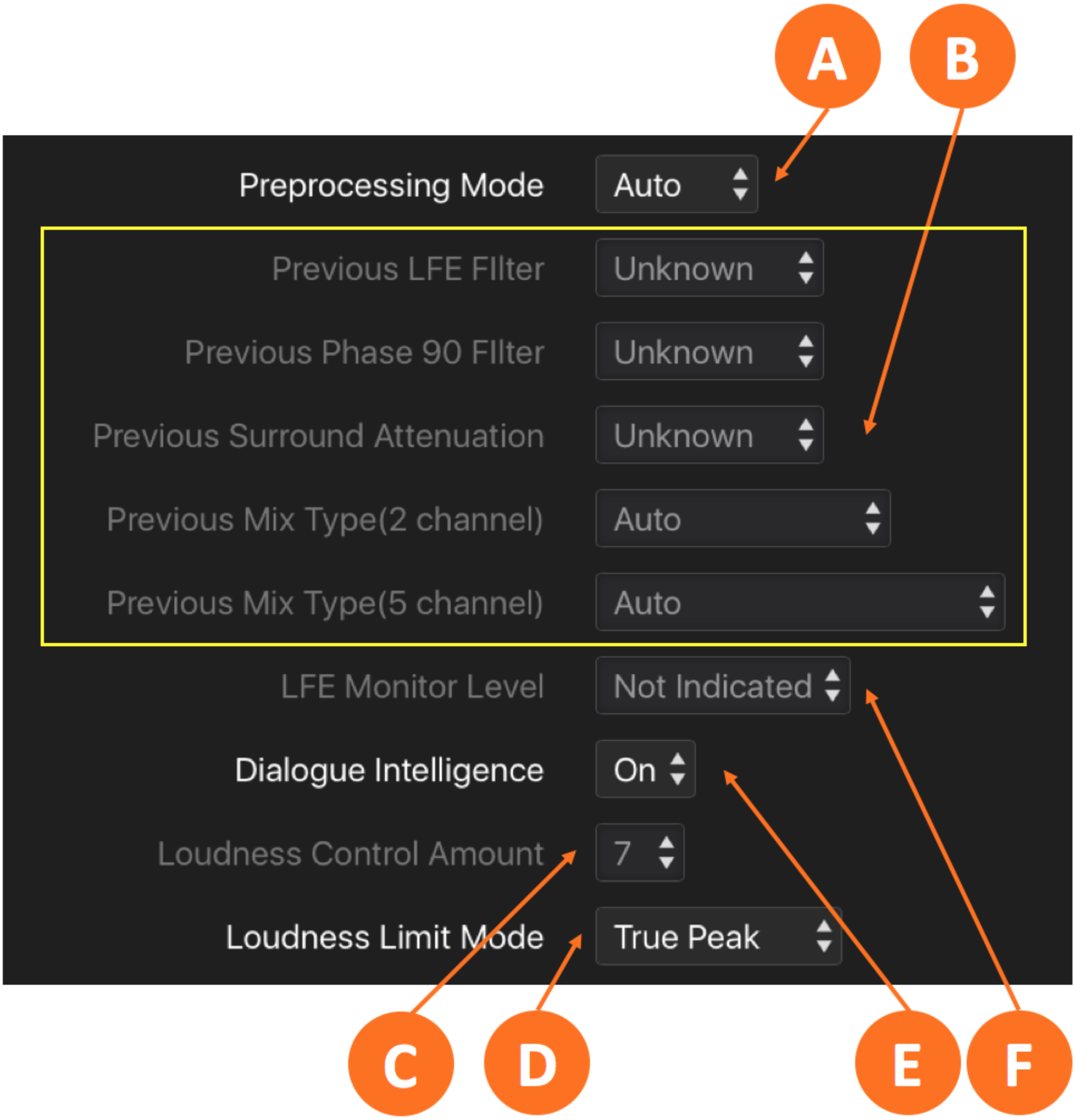


Figure 3 - Bottom portion of Advanced Dolby AC-4 encoder menu

Confidence Monitoring

The LA-5300 includes a Dolby AC-4 decoder for confidence monitoring. Monitor audio is routed to the channels immediately following those used for the bitstream output. For example, if the AC-4 bitstream occupies output channels 1 and 2 and the confidence decoder's channel mode is 5.1, the monitor audio would be present on output channels 3 through 8.

Bear in mind that the controls in the Monitor menu only affect the monitor outputs and not actual bitstream output delivered to downstream facilities or viewers.

Channel Mode

The **Channel Mode control** (4B) sets the channel mode, including 5.1- and 5.1.2-channel surround formats and stereo downmixes in Lo/Ro, Lt/Rt, PLII Lt/Rt, or Headphone.

Target Reference Level

The **Target Reference Level control** (4A) is used to emulate how consumer-side decoders scale the audio. For example, -31dBFS represents Line Mode, whereas -16 or -18dBFS would be useful for mobile-optimized streams.

DRC Enable

Setting the **Enable DRC control** (4C) to "On" applies the DRC calculated by the encoder to the decoded signal.

Dialogue Enhancement Gain

To aid in dialogue intelligibility, the AC-4 codec allows signals that are either flagged with metadata as dialogue or determined to be dialogue by signal analysis to be boosted by the consumer. The **Dialogue Enhancement Gain control** (4E) sets the amount by which dialogue is boosted (or cut).

Dialogue Enhancement Preserve Loudness

When dialogue gain is boosted or cut with the Dialogue Enhancement Gain control, the overall loudness will change compared to the desired target output level. Enabling the **Dialogue Enhancement Preserve Loudness control** (4D) will compensate for this and ensure the overall level matches the loudness target.

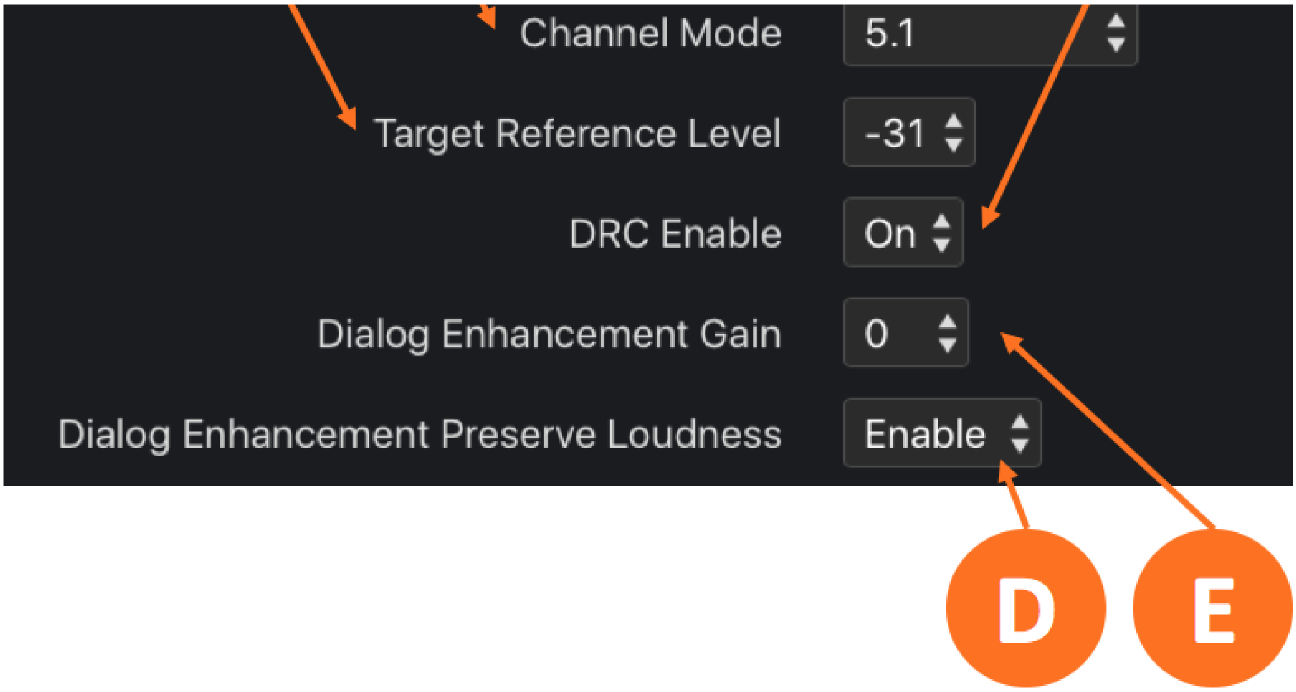
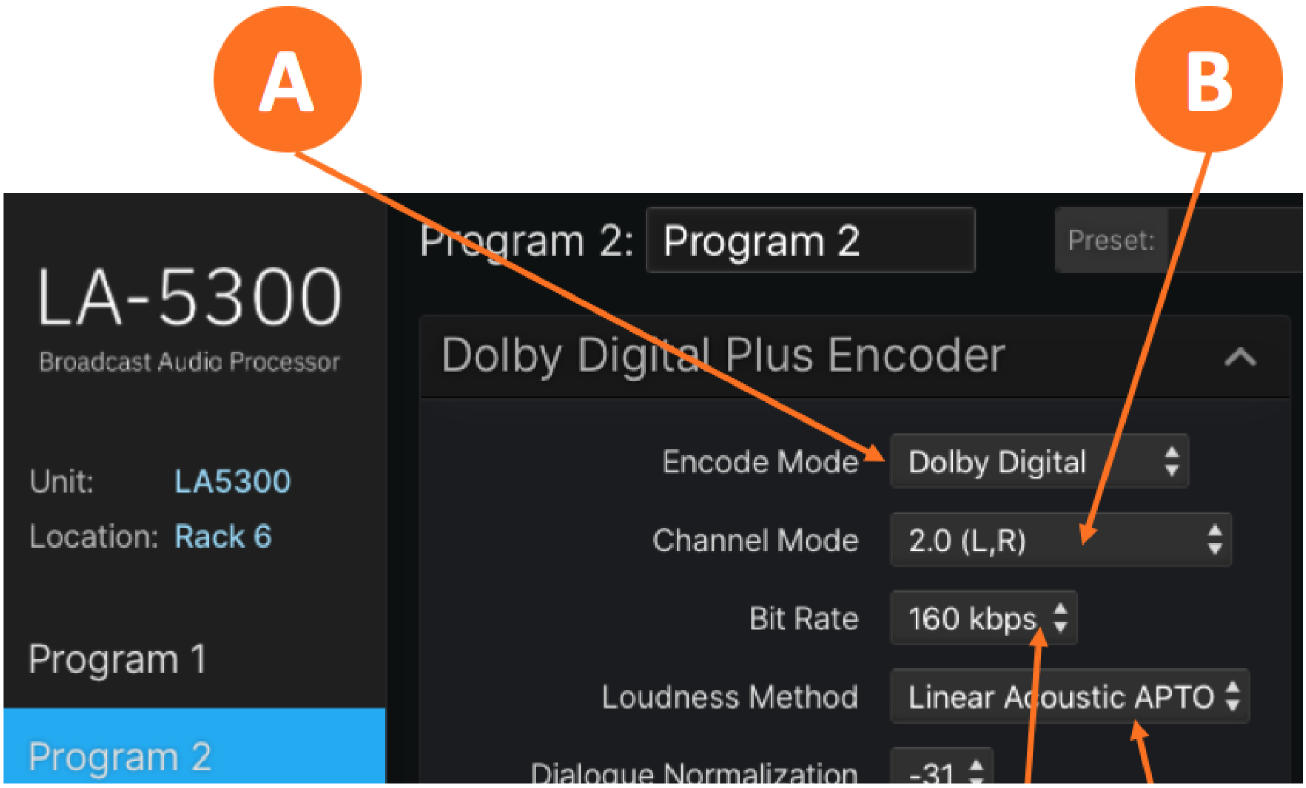


Figure 4 - Monitor menu for AC-4 decoder

Basic Dolby Digital Plus Encoder Menu

Clicking on the Dolby Digital Plus Encoder dropdown menu from the Program 2 screen reveals the basic encoder controls. There is a separate dropdown menu for advanced controls.

Basic parameters include **Encoder Mode** (5A), **Channel Mode** (5B), **Bit Rate** (5D), **Loudness Method** (5E), and **Dialogue Normalization** (5C).



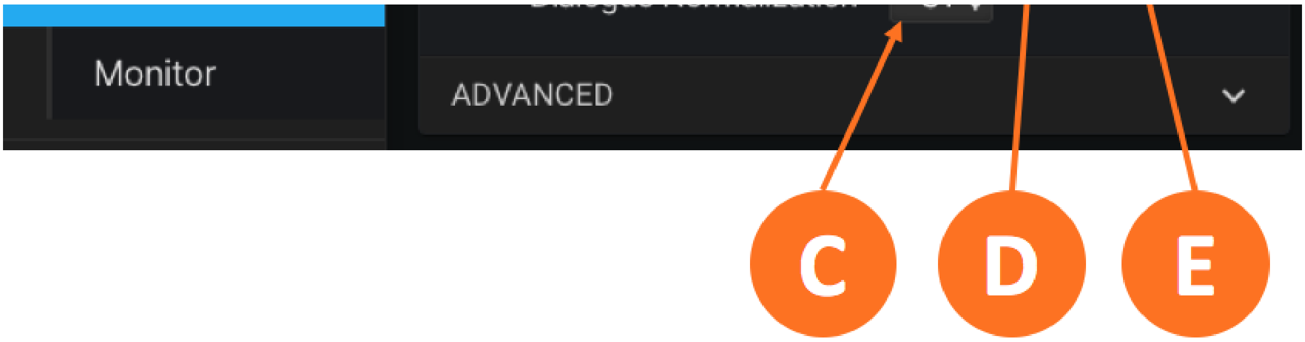


Figure 5 - Basic Dolby Digital Plus encoder menu

Encode Mode

Use the **Encode Mode control** (5A) to choose between Dolby Digital Plus (E-AC-3) or Dolby Digital (AC-3). For ATSC 1.0 applications, Dolby Digital is the proper choice.

Channel Mode

The **Channel Mode control** (5B) determines the channel output mode of the encoder, either 2.0 (L,R) for 2-channel content, or 5.1 (L,R,C,LFE,Ls,Rs) for 5.1-channel content.

Bit Rate

The optimal bitrate for any given application – set by the **Bit Rate control** (5D) - will depend on a number of variables including channel mode, available bandwidth, and desired audio quality. The default bitrates are listed below.

Channel Configuration	Default Dolby Digital (AC-3) Bit Rate	Default Dolby Digital Plus (E-AC-3) Bit Rate
Stereo (2/0)	192 kbps	128 kbps
5.1	384 kbps	192 kbps

Loudness Method

The **Loudness Method control** (5E) determines which processing algorithm is used to achieve an ATSC A/85-compliant output.

Linear Acoustic APTO is the same algorithm used in the Linear Acoustic ARC television processor which provides dynamic range processing and loudness leveling to transparently deliver audio that is both appropriately dynamic (to maintain the creative intent of the program producers) and well-controlled (to ensure CALM-compliance).

The **Dolby Real Time Loudness Leveler**, the same algorithm used in the Dolby AC-4 codec, normalizes the incoming audio signal to achieve ATSC A/85 compliance.

Dialogue Normalization

The **Dialogue Normalization control** (5C) sets the dialnorm metadata value in the Dolby AC-3 or E-AC-3 bitstream, which is the same as the desired output loudness target level. For ATSC A/85, this is normally set to -24dB.

Advanced Dolby Digital Plus Encoder Menu

The advanced parameters menu contains controls that determine whether certain settings and metadata values are set automatically or manually, and, if set manually, the values of individual settings.

Preferred Downmix Method

The **Preferred Downmix Method control** (6A) sets the downmixing metadata value for the downstream decoder.

- **Not Indicated:** No instructions to the downstream decoder are specified, allowing the decoder or the end user to choose the downmix type
- **Lt/Rt:** The surround channels are added in-phase to the Left channel and out-of-phase to the Right channel, allowing a Dolby Surround Pro Logic decoder to reconstruct the Left, Center, Right, and Surround channels
- **Lo/Ro:** The Center and surround channel content is redirected to the Left and Right channels for playback in a system with only two speakers

Line Mode

The **Line Mode control** (6B) sets the DRC (dynamic range control) profile for the line level output on home decoders and set top boxes. It typically uses a profile that uses lighter compression and allows the viewer some latitude in adjusting and scaling the amount of dynamic range as appropriate for their individual listening environment.

RF Mode

The **RF Mode control** (6C) sets the DRC profile for the RF input on a television, usually through the antenna output of a set top box. It typically uses a profile that employs a greater degree of compression and limiting compared to Line Mode, as the RF Mode profile is often used for the “midnight mode” on decoders.

Profiles for both the Line Mode and RF Mode include:

- **Film Standard:** Provides a 5dB null band around the loudness target and offers consistent loudness for most television content while preserving some dynamics
- **Film Light:** Provides a 20dB null band around the loudness target and delivers a much more dynamic and theatrical presentation than Film Standard
- **Music Standard:** Provides a 5dB null band around the loudness target and uses higher ratios than the Film profiles to more effectively manage highly produced music formats

- **Music Light:** Provides a 12dB null band around the loudness target and uses lower ratios to preserve the dynamics of genres such as classical and jazz
- **Speech:** Provides a 5dB null band but higher ratios to more effectively deal with the varying levels and high peaks typically found in speech

Downmix Gain Controls

The **Downmix Gain controls** (6D) can be individually set for the Lo/Ro Surround, Lo/Ro Center, Lt/Rt Surround, and Lt/Rt Center channels.

Values for the Surround channels range from -1.5dB through -6.0dB and –Inf, which disables the Surround channels. Values for the Center channels range from -6.0dB through +3.0db.

LFE Filter

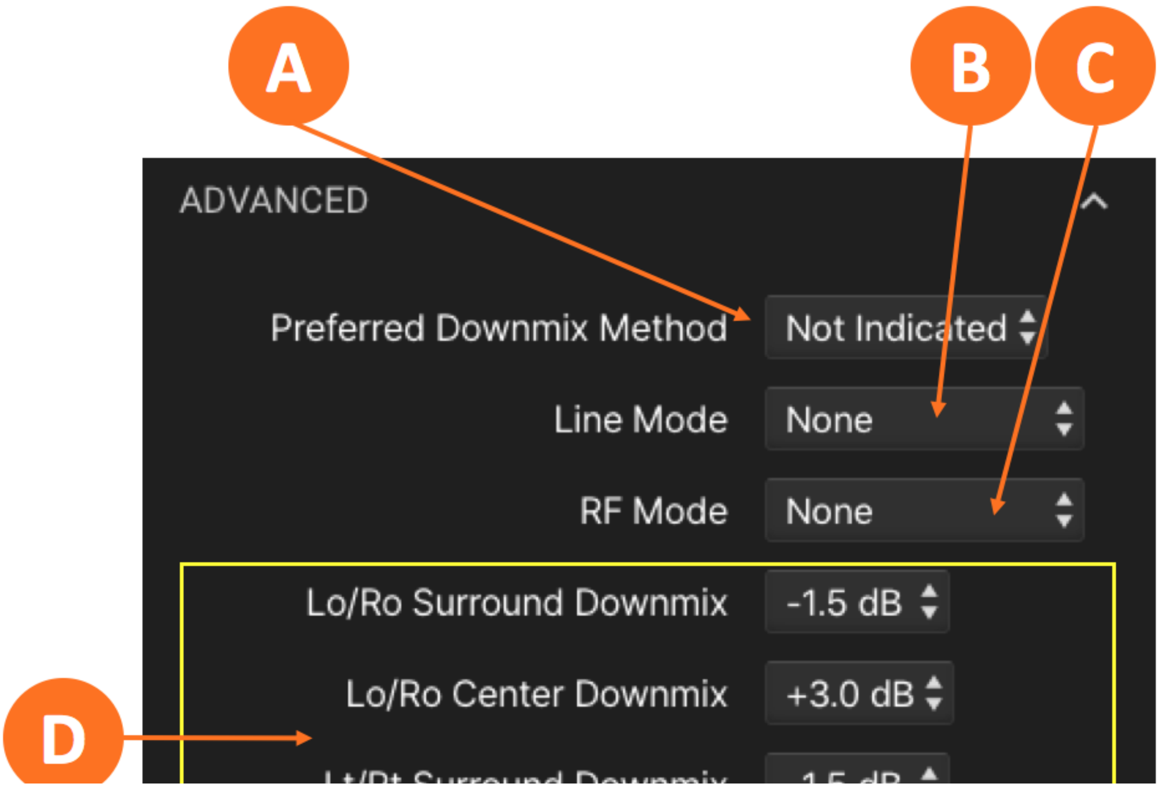
The **LFE Filter control** (6E) applies low-pass filter (LPF) at 250Hz in the LFE channel.

Phase 90 Filter

The **Phase 90 Filter control** (6F) applies a 90-degree phase shift to the Surround channels, allowing the downstream decoder to create a Lt/Rt downmix which can be rendered as an L, C, R, S signal by a Dolby Pro Logic encoder.

Surround Attenuation

Enabling the **Surround Attenuation control** (6G) applies 3dB of attenuation to the Surround channels. This is primarily included for use in cinema applications where multiple surround speakers sum acoustically and can become louder than intended. The recommended setting for broadcast applications is “Disabled.”



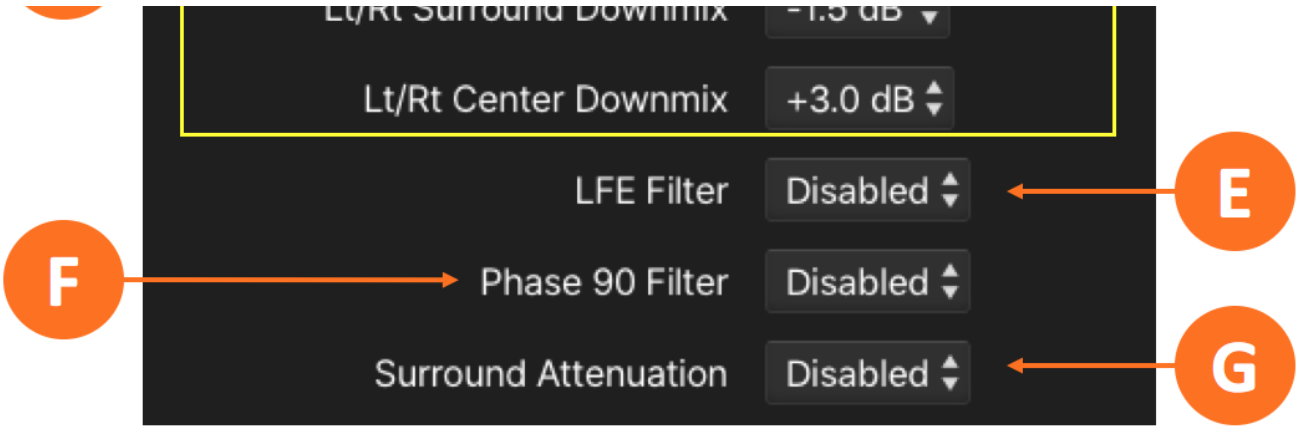


Figure 6 - Advanced Dolby Digital Plus encoder menu

UPMAX® Upmixing

The LA-5300 uses the same new upmixing algorithm found in the Linear Acoustic UPMAX® ISC. If the channel count and format of the incoming content matches the desired output format, the LA-5300 will pass it through as-is. Or, it can upmix stereo and legacy 5.1-channel content to one of several supported immersive formats.

The UPMAX menu is separated into Basic and Advanced sections. The Basic menu is displayed by default. The Advanced menu can be expanded with a dropdown menu as needed.

Basic UPMAX Menu

The controls in the Basic menu can be divided into four basic groups: **Preset and Format controls** (1A), **LFE Channel controls** (1B), **Center Width controls** (1C), and **Auto-Detect controls** (1D).

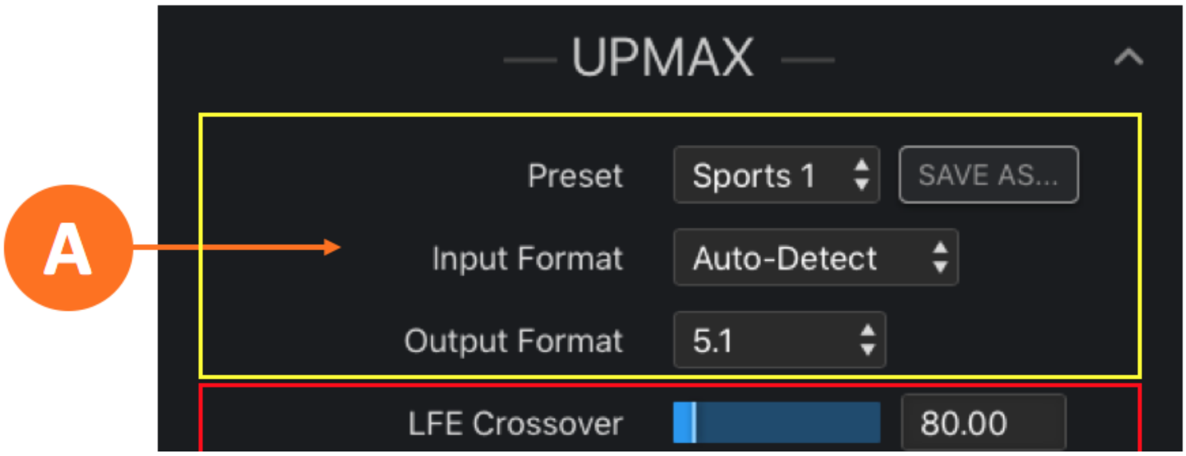


Figure 1 - Basic UPMAX control grouping

Preset and Format Controls

UPMAX processing is largely based on presets. Several factory presets are included which can be used as-is or serve as starting points for a custom user preset.

Preset Menu

Use the **Preset dropdown menu** (2C) to choose a preset. To save any changes as a user preset, click on the **Save As button** (2D) and name the new preset. You must refresh the browser window to see the new preset in the Preset list.

Input Format Menu

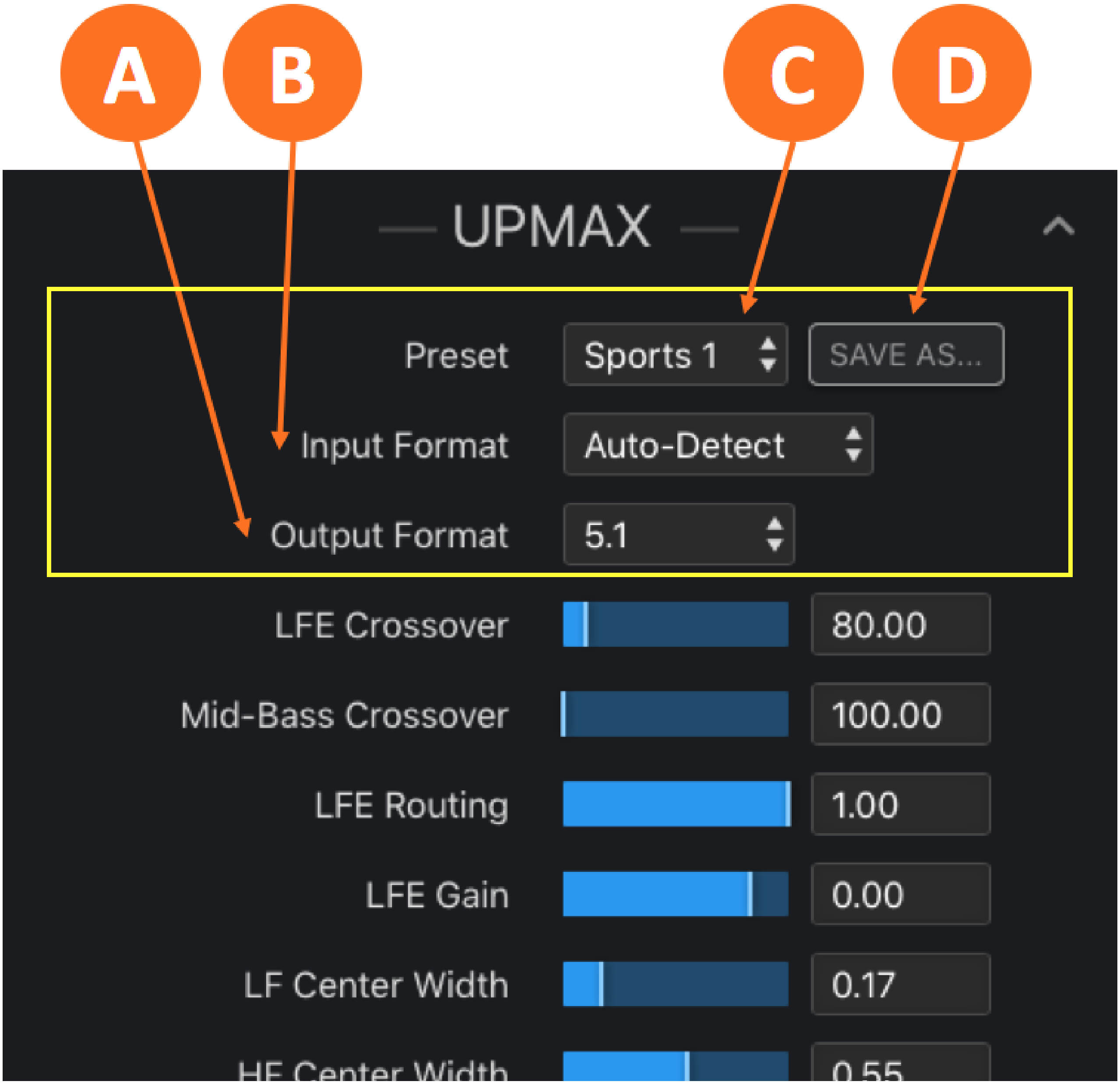
The **Input Format menu** (2B) is used to specify the channel format of the source audio. If the input format matches the selected output format (as selected with the Output Format control) the unit will pass the signal through as-is without additional processing. Input options include:

- **Auto Detect:** In this mode, the unit continuously monitors the input signal to determine its format and will automatically switch to one of the processing modes below (Stereo, Stereo + Dialog, or 5.1)
- **Stereo:** Stereo content will be upmixed to the desired output format
- **Stereo + Dialog:** Stereo content will be upmixed to the desired output format, while a dedicated dialog input on Channel 3 will be passed through to the Center output
- **5.1:** 5-1-channel content will be upmixed to the desired output format
- **Pass-through:** Pass-through will pass the input signal directly through to the output with no additional processing; this can be used as a “soft bypass” function as the signal still runs through the upmixer and therefore incurs the same amount of latency as it would if it was actually upmixing a signal, thereby preserving A/V sync and avoiding lip sync issues

Output Format Menu

The **Output Format menu** (2A) determines the output channel configuration. Output options include:

- **5.1:** Provides a constant 5.1-channel output from Stereo, Stereo + Dialog, and 5.1-channel input sources
- **5.1 Legacy:** Like the standard 5.1 output, it provides a constant 5.1-channel output from Stereo, Stereo + Dialog, and 5.1-channel input sources but with upmixing that is sonically closer to the previous UPMAX algorithm used in the Linear Acoustic UPMAX v4 upmixer and the algorithm employed in the various Linear Acoustic AERO®-series processors
- **7.1:** Provides a constant 7.1-channel output from Stereo, Stereo + Dialog, 5.1-channel, and 7.1-channel input sources
- **7.1+2:** Provides a constant 7.1.2-channel output from Stereo, Stereo + Dialog, 5.1-channel, 7.1-channel, or 7.1.2-channel input sources
- **5.1+4:** Provides a constant 5.1.4-channel output from Stereo, Stereo + Dialog, 5.1-channel, or 7.1.4-channel input sources



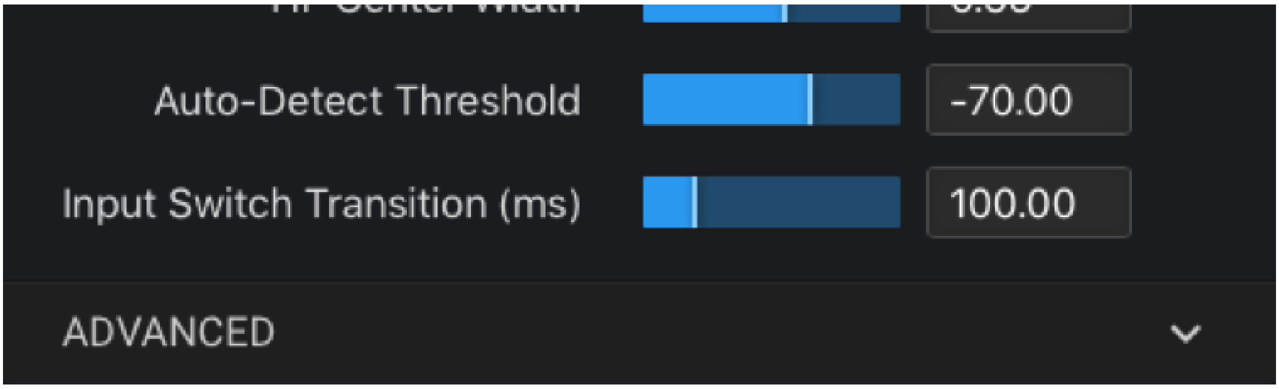


Figure 2 - UPMAX preset and format controls

LFE Channel Controls

The LFE Channel controls are used to tune the crossover frequencies, gain, and signal routing of the low frequency effects audio.

LFE Crossover

The **LFE Crossover control** (3B) defines the crossover frequency at which to extract the low frequency signal routed to the LFE Channel and is typically set between 60 and 100Hz. It works in combination with the LFE Routing and LFE Gain controls to create the LFE Channel.

Mid-Bass Crossover

When a mid-component signal is created for the Center Channel, it is split into two bands, LF and HF. The **Mid-Bass Crossover control** (3A) defines the crossover frequency of the split.

Splitting the mid-component into two bands allows the user to freely control the two signal widths and independently distribute them across the Center, Left and Right Channels.

This can be very useful in keeping the main phase-correlated signal (HF band) in the Center Channel for dialog while spreading the remaining part of the mid-component (LF band) – which could still have some low-frequency dialog along with music and effects – across all front channels.

LFE Routing

The LFE Channel is not typically used in creating a stereo downmix on end-user devices such as set-top boxes (STB) or television tuners. As a result, information in the LFE channel can get lost.

To prevent this, UPMAX uses an internal process that routes some of the extracted low frequency audio to the Center Channel. The amount of low frequency content routed to the Center Channel is set by the **LFE Routing control** (3C). Though values are expressed in a decimal format, they easily correlate to a percentage value (0.00 = 0%, 0.05 = 50%, 1.00 = 100%, etc.).

LFE Gain

The overall gain level of the LFE Channel is set by the **LFE Gain control** (3D) and is typically set at a value of 0dB.

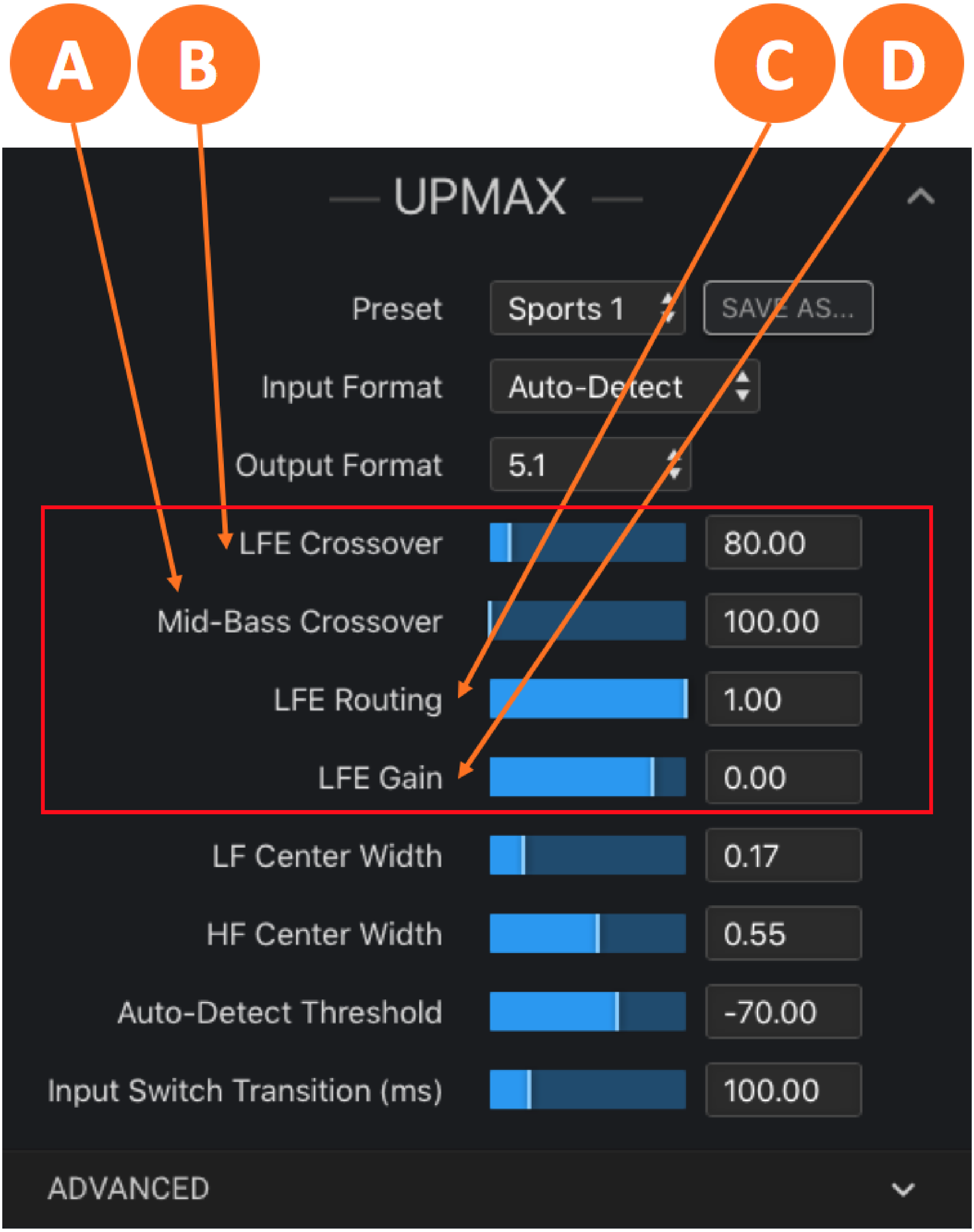


Figure 3 - UPMAX LFE Channel controls

Center Width Controls

These controls define the routing of the LF and HF bands across the Left, Center and Right Channels. Used in combination with the Mid-Bass Crossover frequency control, the provide the ability to craft the image of the Front Channels in a way that maintains a direct sonic-visual association of dialog while providing a pleasant and wide image of the phase-correlated elements. Though values are expressed in a decimal format, they easily correlate to a percentage value (0.00 = 0%, 0.05 = 50%, 1.00 = 100%, etc.).

The **LF Center Width control** (4B) determines the amount of LF audio routed to the Center versus the Left and Right Channels. The **HF Center Width control** (4A) does the same for audio in the HF band.

A setting of 0% sends all audio in the respective band to the Center Channel, while a setting of 100% sends it to the Left and Right Channels only. Intermediate values provide a blend to create any desired mix.

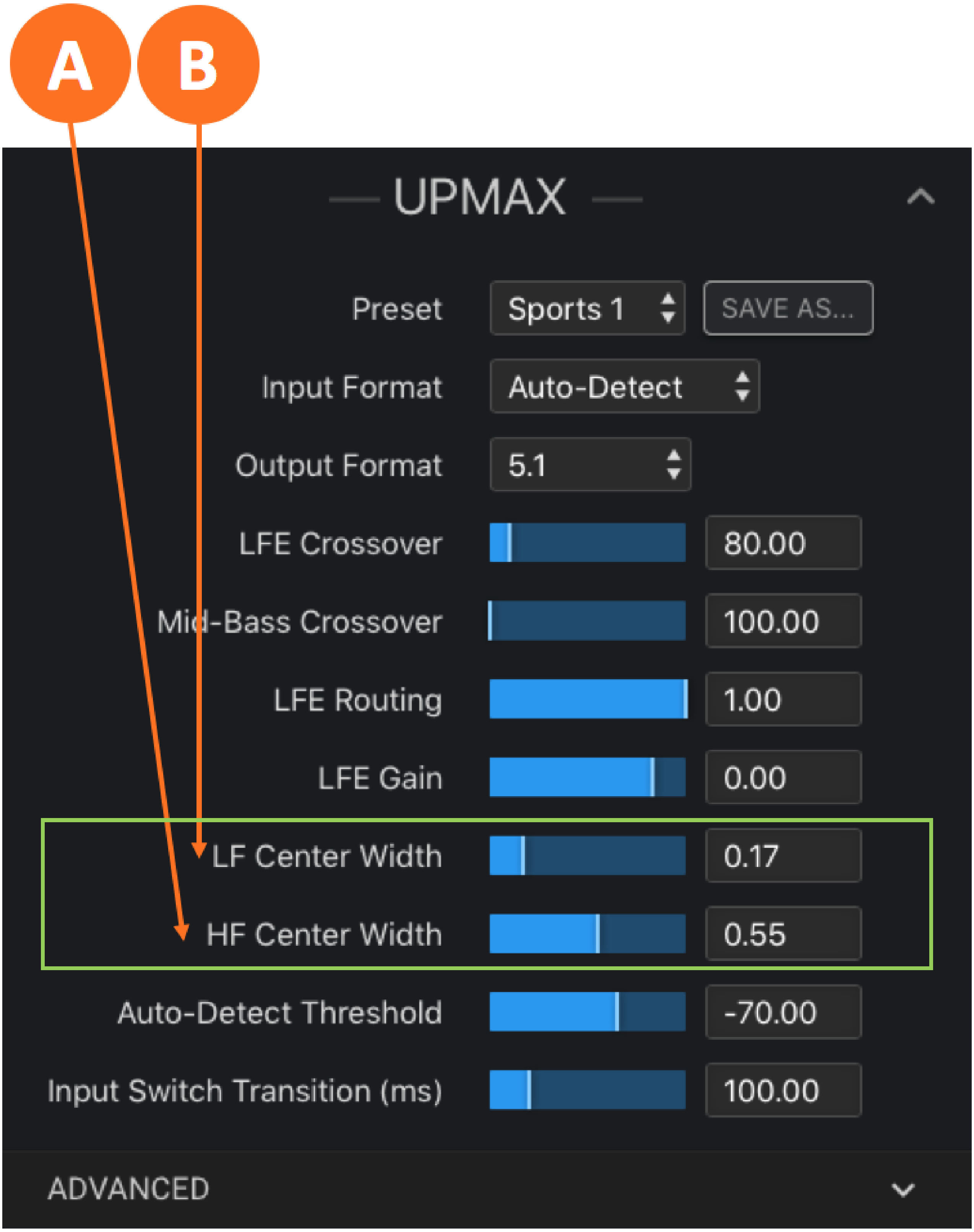


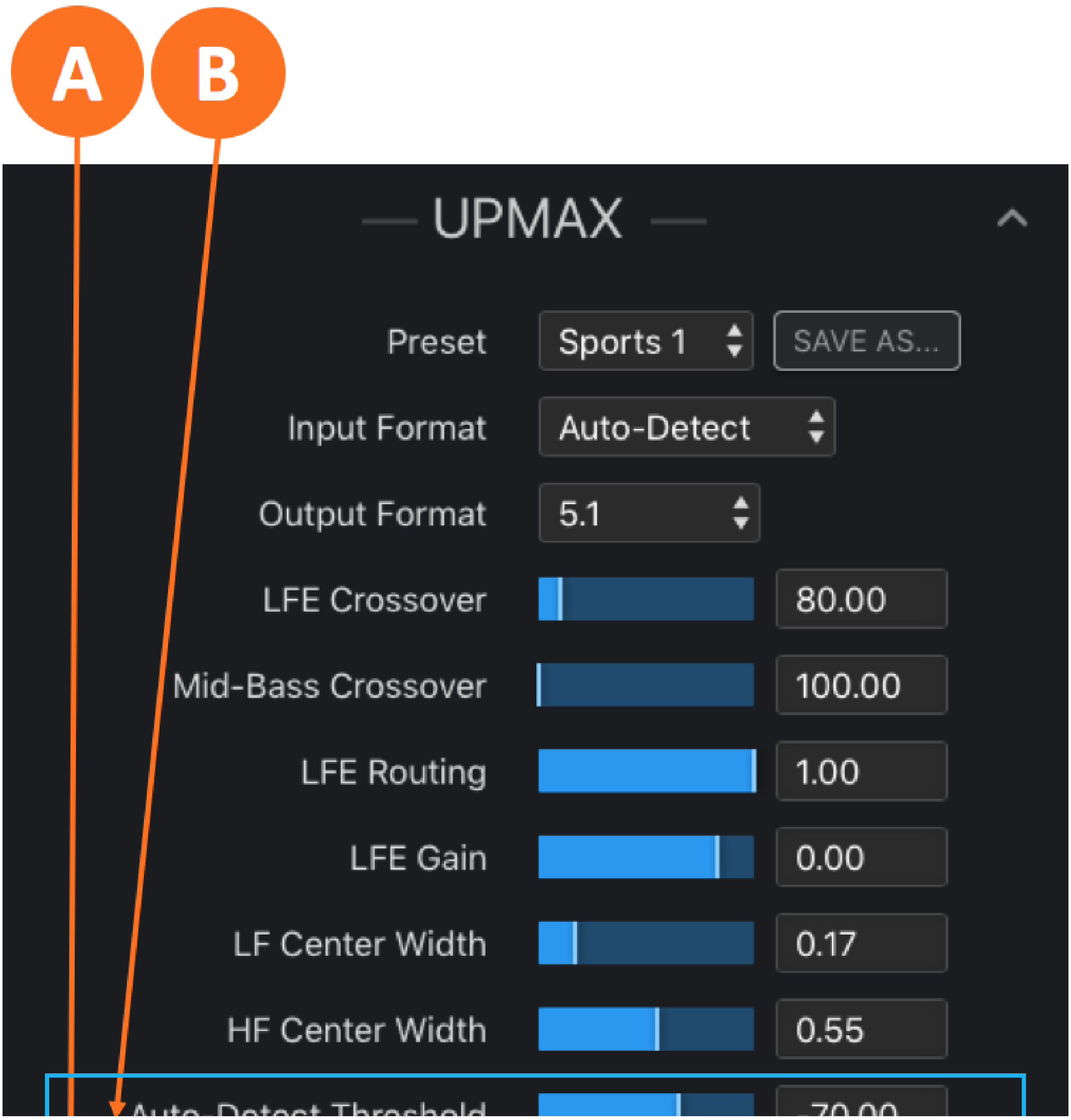
Figure 4 - UPMAX Center Width controls

Auto-Detect Controls

When the Input Format is set to Auto-Detect, the LA-5300 continuously monitors the input signal to determine its format and will automatically switch to Stereo, Stereo + Dialog, or 5.1-channel processing modes.

The level at which the detector circuit determines that the incoming signal is program audio (and not noise or silence) is set by the **Auto-Detect Threshold control** (5B). The proper setting depends largely on the quietness (or noisiness) of channels that are not muted but also contain no audio. Set too low, it may cause a false trigger to upmix; set too high, it may cause a late transition to upmixing. The default value of -70dB should work well for most situations.

The amount of time in milliseconds a signal must stay above the Auto-Detect threshold before the unit begins upmixing - or below the threshold to cease upmixing - is determined the **Input Switch Transition control** (5A). A value of 10ms is a good starting point for most material.



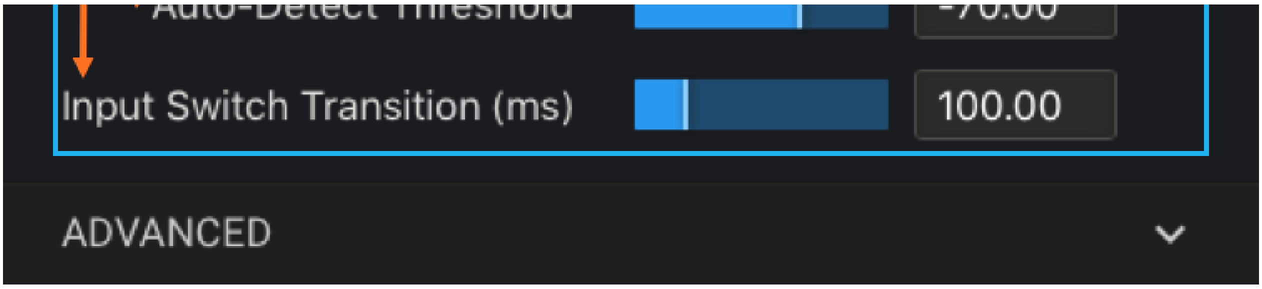


Figure 5 - UPMAX Auto-Detect controls

Advanced UPMAX Menu

The Advanced parameters are included for audio experts who wish to dig deep and tweak settings for very specific applications or demanding tastes. They are not required for most users and are accordingly not displayed by default.

Cycles per Octave

The **Cycles per Octave control** (6A) defines the number of teeth there are in one octave of frequency change of the comb filter. The comb filter is used to separate different frequency components of the input signal to aid in steering sounds in the surround field.

Minimum Comb Filter Frequency

The lowest frequency of the comb filter is set by the **Minimum Comb Filter Frequency control** (6B). For most applications, this should be set at or near the same frequency as the Mid-Bass Crossover frequency.

Comb Filter Level

The **Comb Filter Level control** (6C) sets the overall level of the comb filter. This control can have a significant impact on the movement of audio between the front and rear channels, with higher values rendering a greater sense of depth. In nearly all cases, leaving this control at its default value will result in the most pleasing soundfield.

Front-Rear Balance Factor

The **Front-Rear Balance Factor control** (6D) determines the amount of upmixed sound pushed to the surround channels (analogous to the Surround Depth control in previous UPMAX versions).

Center Channel Gain

The **Center Channel Gain control** (6E) sets the gain of signals determined by the upmixer to belong in the Center Channel, typically dialog.

Rear Channel Downmix Level

In order for a stereo downmix to render at the same level as the original stereo source that was upmixed, the gain applied to the rear channels must be the inverse of the Dolby Rear Channel Downmix Level. Put another way, if you want a 3dB boost in the rear channels, the **Rear Channel Downmix Level control** (6F) should be set to -3.

Front-High Crossover

The **Front-High Crossover control** (6G) adjusts the crossover point between the Front Left and Right channels and the Front Height channels.

Side-Rear Filter Crossover

The **Side-Rear Filter Crossover control** (6H) adjusts the crossover point between the Left and Right Surround channels and the Left and Right Back channels when upmixing to 7.1.

Side-to-Rear Gain LF

The **Side-to-Rear Gain LF control** (6I) adjusts the gain of the low frequency content in the Surround and Rear channels when upmixing to 7.1.

Side-to-Rear Gain HF

The **Side-to-Rear Gain HF control** (6J) adjusts the gain of the high frequency content in the Surround and Rear channels when upmixing to 7.1.

Rear Channel Boost

The gain of the Left and Right Back channels when upmixing to 7.1 is set by the **Rear Channel Boost control** (6K).

Highs in Laterals

The **Highs in Laterals control** (6L) determines to what degree higher frequencies from the ear-level to height speaker crossover circuit are mixed back into the ear-level speakers.

Lows in Heights

The **Lows in Heights control** (6M) determines to what degree lower frequencies from the ear-level to height speaker crossover circuit are mixed back into the height speakers.

Ceiling Speaker Gain

The **Ceiling Speaker Gain control** (6N) determines the amount of gain added to the Height channels.

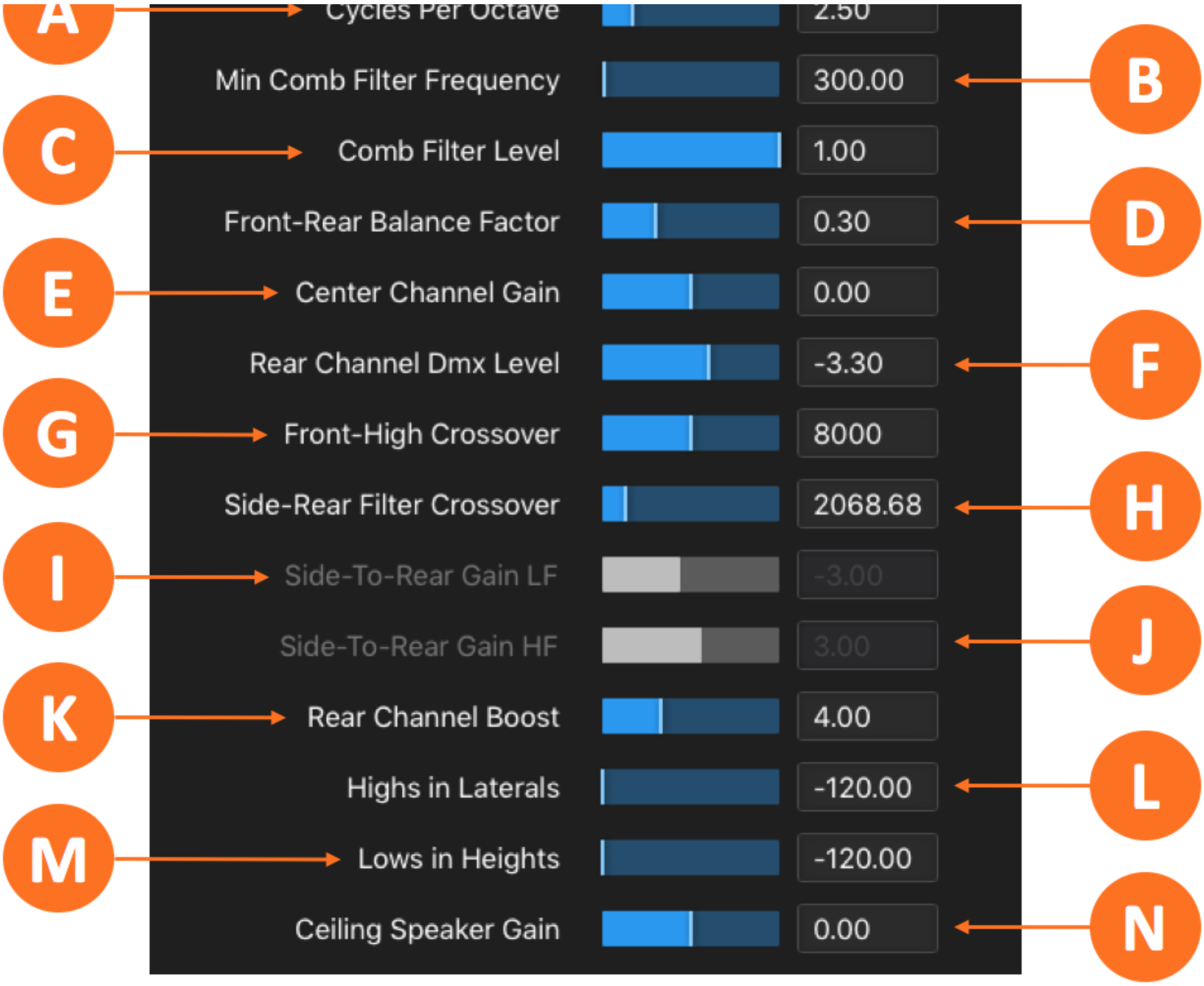


Figure 6 - UPMAX Advanced menu controls

Nielsen® and Verance® Watermark Encoding

The LA-5300 includes optional support for audience measurement watermarking from both Nielsen and Verance.

In both applications, proprietary and customer-specific information is required and is provided by the vendor.

Nielsen Watermark Encoding

Set the **Nielsen Watermark Enable control** (1B) to “On” to enable the encoder.

The **Active Process control** (1A) determines which watermarking type or which combination of types is generated. Please check with Nielsen to determine the correct process for your station. Supported processes include N2, N6, N2+N6, CBET, CBET+N6, and CBET+N2+N6.

Use the **SDI, Check Digit, CSID, and CBET Check Digit fields** (1C) to enter the proper codes as provided by Nielsen for your individual station. The Status window will indicate whether or not the watermark is being properly encoded.

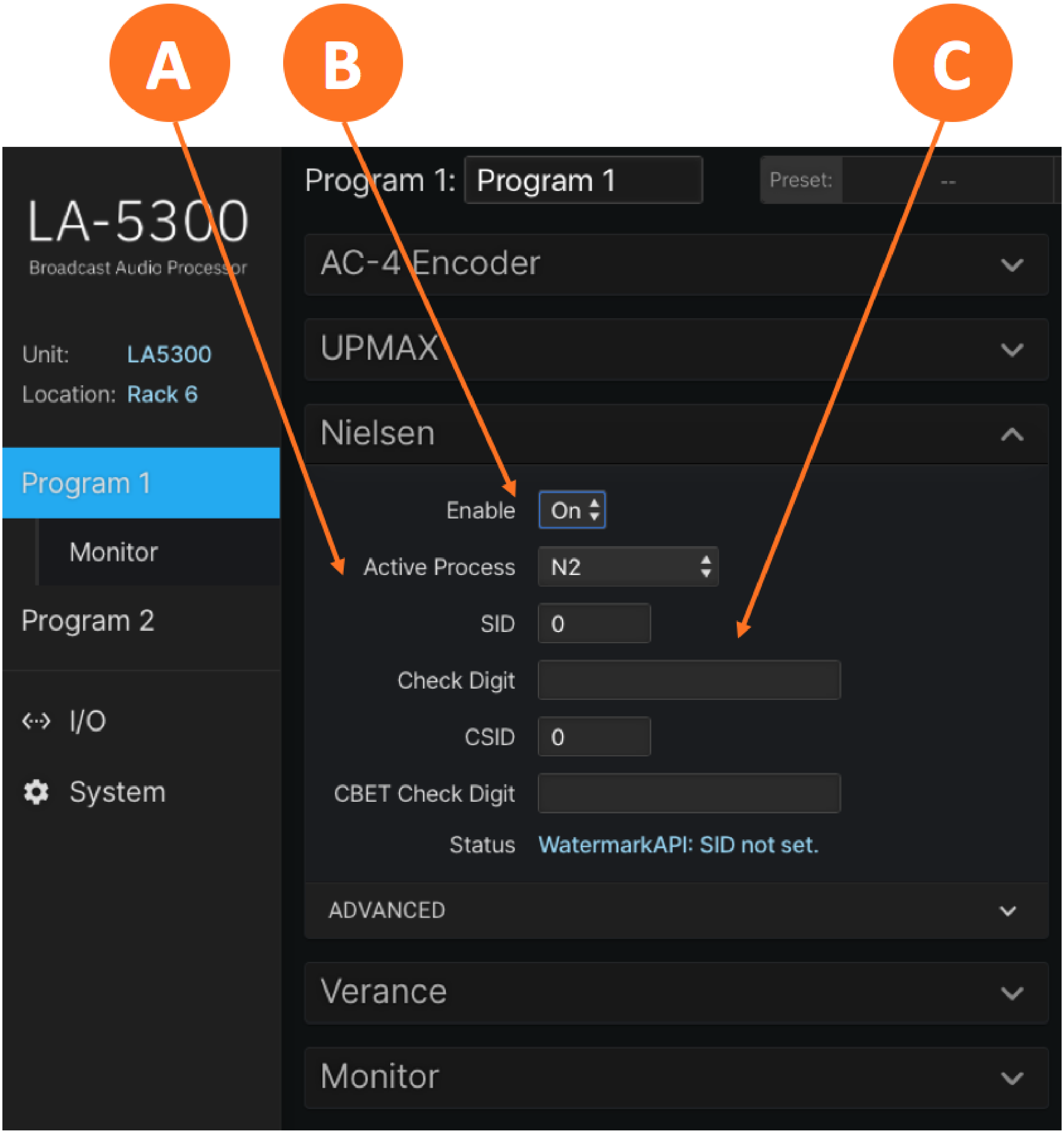


Figure 1 - Nielsen watermark encoding menu

Verance Watermark Encoding

Set the **Verance Watermark Enable control** (2B) to “On” to enable the encoder. Check the **Status window** (2C) to verify that the encoder is running.

The **Calibrate control** (2A) is used in conjunction with a special audio test file to measure delay when the Verance watermark is inserted. The file and specific instructions are provided to customers by Verance.

Use the **Server Code** and **Interval Code fields** (1D) to enter the proper codes as provided by Verance for your individual station.

Clicking on the **Advanced dropdown menu** (1E) will display the current Verance library version.

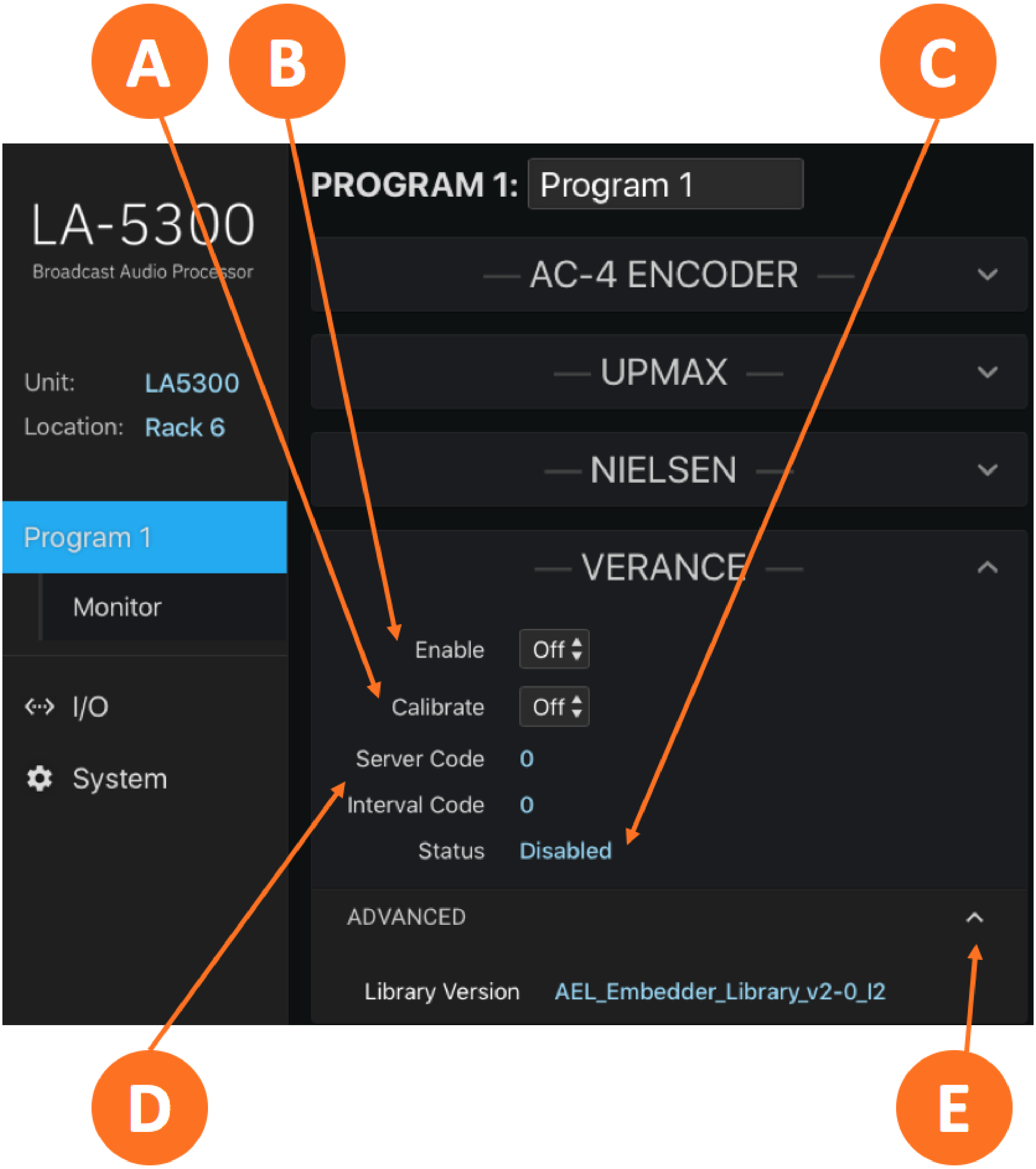


Figure 2 - Verance watermark encoding menu

APTO® Processing

LA-5300 units equipped with the Dolby Digital Plus Encoder for a second program source offer a choice of loudness control options including Dolby's own Real Time Loudness Leveler (RTLL) and Linear Acoustic® APTO processing.

There are no dedicated controls when using the RTLL option, but APTO is a bit more complex and requires some additional explanation.

How Does APTO Work?

In very basic terms, APTO first measures and analyzes the loudness of the incoming audio. In its first processing stage – the "Dynamic Range" stage - it applies realtime loudness control to reduce the overall dynamic range and get the levels within the user-defined comfort zone. This processed audio is then scaled in the second processing stage – the "Compliance" stage - to achieve an average output level that matches the desired target loudness value.

Basic APTO Controls

The individual controls within each APTO factory profile have been adjusted in such a way that they automatically deliver the required result associated with the profile name, such as "EBU R128 Adaptive" for use in most European regions and "ATSC A/85" for CALM-compliance in the U.S. However, there may be situations in which you want to adjust certain individual parameters to customize or fine tune the audio.

Current Profile

The currently selected profile is show in the **Profile window** (1J), which also serves as the dropdown menu for selecting a profile.

Target Loudness

The **Target Loudness control** (1A) sets the desired average loudness level of the output signal in either LUFS or LKFS, depending upon the profile. Some profiles, such as EBU R128, will measure loudness according to overall program levels while others, such as ATSC A/85, will do so based on dialog and gated speech measurements.

Adaptation

The **Adaptation control** (1B) determines how much processing is applied to the incoming signal in the Dynamic Range processing stage, and, in combination with various individual controls, determines dynamic range of the output audio.

The ideal amount of Adaptation depends upon both the source content and on the destination platform. Content that has been pre-analyzed for loudness, scaled, and normalized in the file domain will require less realtime processing than, say, live sports, which can have rather unpredictable audio levels. Programming streamed to a mobile device or expected to be heard on lower quality earbuds will benefit from more Adaptation than the same content destined for a home cinema presentation.

Ideally, you want enough Adaptation to achieve compliance and keep levels within the viewer's comfort zone and at a stable average level, but not so much that the audio sounds unnatural or over-processed. APTO is based on a psychoacoustic model that takes into account human hearing and perceived loudness and remains very natural-sounding even when extensive processing is applied, but it is generally advisable to keep the amount of Adaptation under 50% when possible. If incoming content is so poorly controlled as to require higher values, it may be necessary to adjust some of the individual controls and create a custom profile to address this scenario.

Bypass

APTO can be bypassed so that the input audio is passed through to the output without being processed by clicking on the **Bypass control** (1C). This is useful for a quick comparison between the unprocessed and processed audio.

Reset

Clicking on the **Reset butt**on (1D) resets the loudness measurements as well as the gain buffers APTO uses in the normalization stage. Resetting at the start of each individual program element provides accurate per-segment loudness measurements, aids in achieving overall compliance, and ensures that adaptive processing decisions are made based upon the current program dynamics. A GPIO input may be used to trigger the reset automatically.

Maximum True Peak Limiter and Limiter Threshold

The **True Peak Limiter control** (1E) enables and disables the True Peak Limiter, which is the final processing stage just ahead of the final output. The **Maximum True Peak value** (10-1F) sets the level beyond which the True Peak limiter engages and attenuates the processed audio so as not to exceed the set level. These controls comply with the True Peak measurement as outlined in ITU-R BS.177-4 Annex 2.

Adaptive Input Detection

When enabled, the **Adaptive Input Detection control** (1G) dynamically adapts the amount of processing occurring in the Dynamic Range stage of processing depending upon the actual measured average level at the input. The degree to which the actual input levels influence the processing versus relying upon the value of the **Average Input Level control** (found in the Advanced menu) is determined by the **Adaptive Input Percentage control** (also located in the Advanced menu).

When **Adaptive Input Detection** is disabled, Dynamic Range processing decisions are made based strictly upon the value set in the **Average Input Level** (found in the Advanced menu) and in accordance with the settings of other parameters and controls.

Adaptive Input Detection is especially useful when source audio levels are unknown or are likely to vary widely as it allows the Dynamic Range processing stage to respond more predictively the actual incoming content.

If the incoming content has already been analyzed and loudness-corrected in the file domain, less overall processing is required and a more natural-sounding output can be achieved by setting the **Average Input Level** to the same value as the target level used during file-based correction and reducing the value of the **Adaptive Input Percentage** control (also found in the Advanced menu) or disabling **Adaptive Input Detection** altogether.

Dialog Normalization

The **Dialog Normalization control** (1H) enables dialog detection and measurement. When enabled, APTO's Compliance processing stage uses long-term speech-only measurements rather than the overall input program loudness to ensure the output target level is achieved.

The use of dialog-based measurement is normally determined by regional regulations. For example, ATSC A/85 relies upon anchor-based normalization, specifically dialog, while EBU R128 recommends the use of overall measurements. This is reflected in their respective factory profiles.

Average Hold

When the **Average Hold control** (1I) is enabled and the output audio of the Dynamic Range processing stage falls below the level set in the **Average Hold Threshold control** (located in the Advanced menu), APTO's Compliance stage of processing becomes inactive until the level once again rises above the threshold. This helps prevent noise and low-level background audio from being increased unnecessarily.

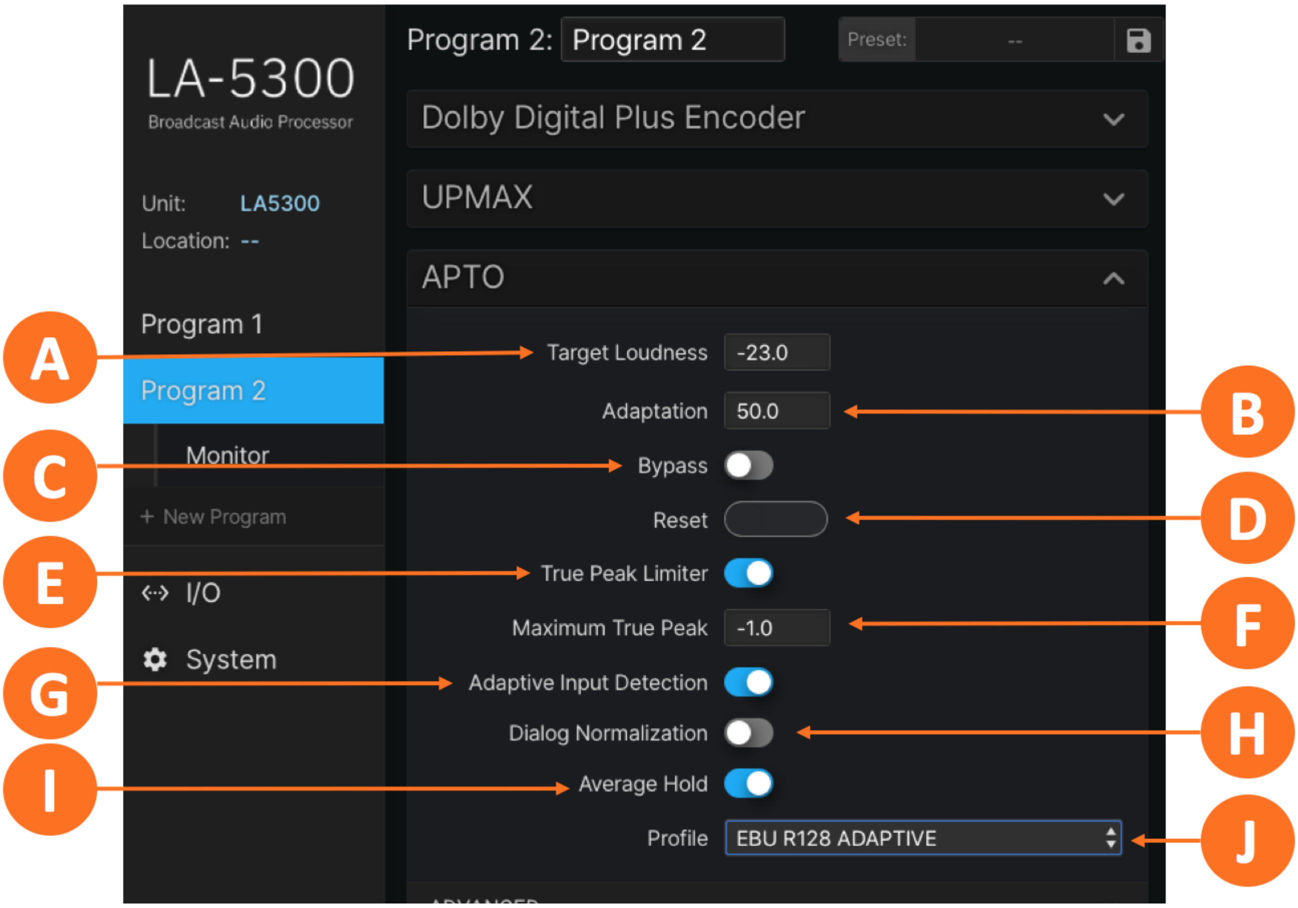


Figure 1 - Basic APTO controls

Advanced APTO Controls

As we’ve mentioned before, most users and applications will be well served by simply selecting the appropriate factory profile according either to compliance regulations. At most, some adjustments to the Basic controls may be necessary.

Unless there are extenuating circumstances or the need to solve a particular problem with highly difficult programming, adjusting the Advanced parameters described below is rarely necessary.

> **Important!** Making arbitrary adjustments without a firm understanding of how these controls work, relate to the Basic APTO controls, and interact with the complex processing algorithms can cause more harm than good and result in audio that is either unpleasing, non-compliant, or both.
>
> That said, they are available for those who need them.

Audio High Pass Filter Cutoff Frequency

The **High Pass Filter control** (2A) sets the cutoff frequency for the high pass filter applied to the input signal. No audio below this frequency will make it through to the APTO processing engine. In most cases, this control should be set to “Off”, but in some instances, it may be beneficial to filter out any sub-audible frequencies (below 20Hz).

Voiced-based content with no low frequency information can benefit from a higher setting (60Hz, for example). Delivery platforms with limited headroom (such as in-flight entertainment or mobile streaming) and consumer devices with small speakers (mobile phones, tablets, or earbuds) can also benefit from a higher setting to prevent distortion.

Average Hold Threshold

When the **Average Hold control** (found in the APTO basic menu) is enabled, the **Average Hold Threshold control** (2B) determines the point at which the hold is triggered in the Compliance processing stage to prevent increasing noise or background audio.

The value at which the control is set is relative to the **Target level**. For example, if the **Target level** is set to -23dB LUFS and the desired level at which the hold engages in the Compliance processing stage is -34dB, then the **Average Hold Threshold** should be set to -10dB.

Minimum Speech Duration

Proper dialog-based loudness processing requires the accurate detection and measurement of speech. When **Dialog Normalization** is enabled, the **Minimum Speech Duration** control (2C) determines how

many seconds of continuous speech are required for the measurements to be considered reliable and valid, and after which the Compliance processing stage adaptively switches to dialog normalization mode.
A minimum of 5 seconds is recommended. Shorter values may incorrectly factor in non-dialog material, and significantly longer values may cause unnecessary delays in engaging dialog normalization processing, especially in content with shorter segments of continuous dialog.

Dialog Compliance Measurements

The **Dialog Compliance Measurements** control (2D) determines how many speech measurements are used to perform dialog normalization in the Compliance stage of processing. Because the algorithm samples the measurements at 0.5 second (one half of one second) intervals, this control should be set to a time value that is twice the desired average. For example, to base normalization on a 60 second average, the control should be set to a value of 120.

> ⓘ **Note -** The Dialog Compliance Measurements setting only applies to profiles that use Dialog Normalization. Profiles that **do not** use Dialog Normalization rely upon the settings of Compliance Window control to determine their time window.

Adaptive Input Percentage

When the **Adaptive Input Detection control** (located in the Basic section) is enabled, the Dynamic Range processing section of APTO dynamically adapts the amount of processing depending upon the actual measured average level at the input. The degree to which the actual input levels influence the processing versus relying upon the value of the **Average Input Level control** (2L) is determined by the **Adaptive Input Percentage control** (2E).

The recommended setting to start is 50% in order to keep the effect of adaptation more consistent.

Foreground Sounds Coefficient

All programming has what is sometimes referred to as an “anchor element,” that is, the audio content to which the viewer will pay the most attention. This is typically (though not always) dialog. This is also referred to as “foreground” audio to differentiate it from background audio, or in some cases, noise.

The **Foreground Sounds Coefficient control** (2F) sets the level in the Dynamic Range processing stage at which program audio is no longer considered a foreground sound relative to both the lower border as set by the **Null Area Coefficient** control (2G, described in detail below) and a minus infinite level (full silence) on a proportional scale.

For example, if the **Target level** is set to -23dB LUFS, the **Null Area Coefficient** is set to 4, and the **Foreground Sounds Coefficient** is set to 2, any audio between -27 and -25dB will be deemed foreground audio and therefore be raised toward the target.

Audio at levels lower than -27dB will still be increased toward the target, but the degree to which gain is increased slows down considerably. The lower the audio is from -27dB, the less the gain will increase.

Null Area Coefficient

The primary goal of APTO processing is to deliver a consistent *average* output level as set by the Target Level control. This does not mean, however, that the actual output audio level must never deviate from this value. In fact, a certain amount of dynamic range helps preserve the artistic integrity of the original programming and makes for a more engaging audio experience for the viewer.

One of the things that makes APTO different from traditional processing is its ability to “do nothing” to audio levels when no action is required to maintain the correct average output level. This avoids the “busy” sound of traditional compressors and ACGs which by their very nature are always operating either over or under a threshold, and therefore always increasing or decreasing gain - often for no good reason.

The **Null Area Coefficient control** (2G) sets the lower and upper thresholds that together determine the size of the window in which APTO’s Dynamic Range processing stage neither increases nor decreases gain, with the user-determined Target level sitting in the middle of the range. Values are in dB, with larger values resulting in a larger “do nothing” window.

Compliance Speed

How quickly gain changes are made in the Compliance processing stage are largely program dependent. However, the maximum rate at which the gain can increase or decrease is set by the **Compliance Speed control** (2H). The rate is calibrated in dB (or LU) per second.

Higher values (from 2 – 6 LU per second) allow compliance to be achieved more quickly but may introduce audible gain changes to the average program level in the process. These faster settings are best suited for situations when input levels are expected to be very inconsistent.

Lower values (from 0.2 to 2 LU per second) provide a subtler and more natural-sounding normalization but may result in gain changes that are too slow and allow the audio to remain outside of the comfort zone for too long. Slower settings work well for content that is more consistent and well-controlled. They are also recommended for long-form content such as feature films or classical music.

Average Maximum Gain

The **Average Maximum Gain control** (2I) sets the maximum amount of positive gain (gain increase) applied in the Compliance processing stage in order to reach the Target level.

Larger values will allow very soft program segments to be raised by a greater amount, but whether or not this is desirable must be considered. For instance, some material may have been purposefully kept at a lower level for dramatic effect, and of course there is always the risk of increasing unwanted background noise.

Another unwanted byproduct of setting this control too high - especially with profiles that use a lower **Adaptation value** – is that it can result in APTO taking too long to lower levels when loud content immediately follows soft content, such as when a loud commercial follows a quiet passage from a TV drama, as it will take longer to reduce a greater amount of gain.For these reasons, it is advisable to set the **Average Maximum Gain** for no more than 2 LU.

One way to enhance the efficiency of the processing across programs is reset APTO at the transition point, which will in turn reset the amount of average gain or attenuation being applied and allow each individual program element to be optimally processed from the beginning.

Average Maximum Attenuation

The **Average Maximum Attenuation control** (2J) sets the maximum amount of negative gain (gain reduction) applied in the Compliance processing stage in order to reach the Target level.

Larger values will allow loud program segments to be reduced by a greater amount. However, setting this control too high – again, especially with profiles that use a lower **Adaptation value** - can cause APTO to take too long to raise levels when soft content immediately follows something loud, such as when transitioning from a loud commercial back to a quiet TV drama, as it will take longer to boost a greater amount of gain.

Just as with the **Average Maximum Gain control**, resetting APTO at the start of each program segment can help minimize such issues and optimize overall processing for each individual program segment.

Compliance Window

The **Compliance Window control** (2K) adjusts the size of the sliding time window used in the Compliance processing stage to align the output program to the target level. It is similar to the rolling integration time on an LKFS/LUFS loudness meter.

> **Note -** The Compliance Window setting only applies to profiles that do **not** use Dialog Normalization. Profiles that **do** use Dialog Normalization rely upon the settings of the Dialog Compliance control to determine their time window

Many factors can influence whether or not content at the output is compliant. These include the level consistency of the incoming audio, program duration, the difference between the average input level and the Target level, the amount of Adaptation employed, and the settings of many of the controls listed in this chapter.

That said, a smaller compliance window value more easily delivers a compliant output for shorter program durations, but can produce more variations in long-term programming.

On the other hand, a larger compliance window value will allow for a smoother and more consistent average overall for longer program durations, but can reduce the effectiveness of short-term normalization, especially program elements shorter in duration than the compliance window.

For short-term programs, a value in the range of 20-30 seconds is recommended. Larger values, from 60-120 seconds, are recommended for mid- to long-form content. For profiles which rely upon Dialog Normalization, a setting between 120-180 seconds is highly recommended.

Average Input Level

The value of the **Average Input Level control** (2L) should be set to match the average level of the input audio as it serves as the mid-point value of the comfort zone defined by the **Null Area Coefficient** control.

For content that has been previously analyzed and loudness corrected in the file domain, this value will be easy to determine.

For live programming, unprocessed material, or in cases where programming is of different genres from different decades, this becomes more challenging. For example, cinematic content is typically mixed to -31/-27 LUFS, while music productions can be as loud on average as -12/-8 LUFS.

In these cases where the average input level is not easy to predict, it is best to enter a value that matches the output target loudness value, enable **Adaptive Input Detection**, and take advantage of APTO's intelligent dynamics processing.

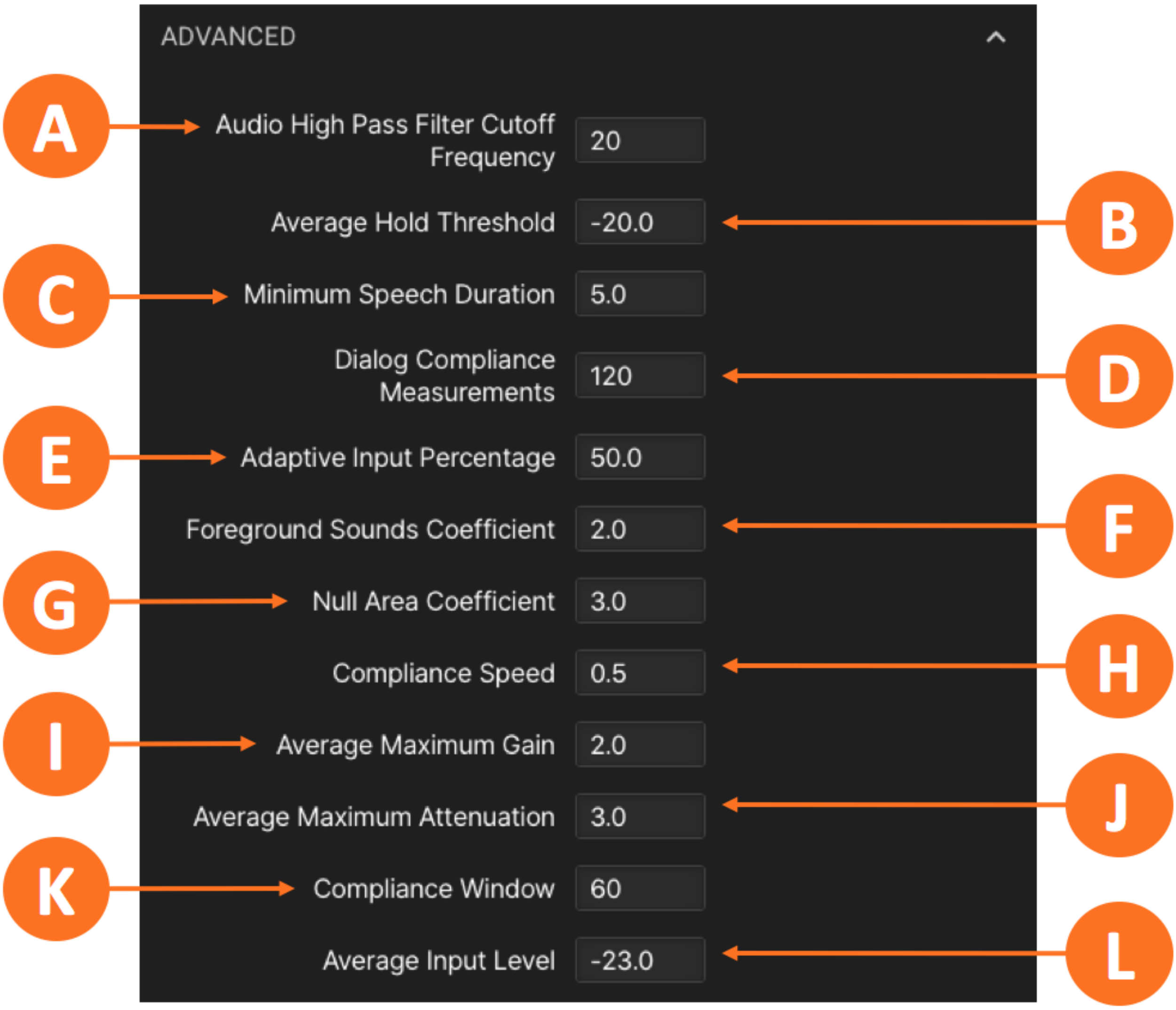


Figure 2 - Advanced APTO controls

Specifications

Processing

Loudness Control

- Program 1 (Dolby AC-4 encoding) – Dolby Real Time Loudness Leveler (RTLL)
- Program 2 (Dolby Digital Plus encoding) – Dolby Real Time Loudness Leveler (RTLL) or Linear

Acoustic APTO™

Upmixing

- Linear Acoustic UPMAX® ISC upmixing

Decoding

- Decodes Dolby Digital (AC-3), Dolby Digital Plus (E-AC-3), and Dolby AC-4 at the input for watermarking, bitstream analysis, and monitoring
- Supports Dolby AC-4 pass-through

Transcoding

- Transcodes Dolby Digital and Dolby Digital Plus to Dolby AC-4

Encoding

- Encodes to Dolby AC-4 with Dolby Real Time Loudness Leveler (RTLL)
  Optional encoding of second program to Dolby Digital Plus (E-AC-3) or Dolby Digital (AC-3) with Dolby Real Time Loudness Leveler (RTLL) or Linear Acoustic APTO® processing

Audience Measurement Watermarking

- Optional Nielsen and Verance audience measurement watermark encoding

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I/O

SDI I/O

- 3Gb/s HD/SD-SDI input (SMPTE ST 425-1, 292M, and 259M), up to 1080p/60/59.94/50Hz
- Optional Quad-Link 3Gb/s SDI for 4K workflows (mutually exclusive with MADI option)

AES-3 I/O

- Five stereo pairs via 75 Ohm BNC unbalanced female connectors, internally terminated; signal levels per SMPTE 276M/AES-3ID-2001

AES67 I/O

- 16 channels of bi-directional AES67 I/O in support of SMPTE ST 2110-30 and -31 workflows

MADI I/O

- Optional MADI I/O supports up to 16 channels for processing (mutually exclusive with Quad-Link SDI option)
- Passthrough and shuffling for up to 64 channels
- I/O via coax or optical SFP socket (SFP sold separately)

Latency

- AES-3 input to AES-3 output with SRCs disabled: 30.5ms
- AES-3 input to AES-3 output with AES-3 input SRC enabled: 33.5ms
- SDI input to SDI output with SRCs disabled: 33.5ms
- SDI input to SDI output with SRCs enabled: 36.5ms
- AES67 input to AES67 output: 107ms (measurement made using Telos Alliance SDI xNode which has a latency value of 80ms)
- Latency is subject to change depending upon software version

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Reference

- 48kHz reference via SDI, PTP, AES-3, internal clock (standalone use only), or MADI (when MADI option is installed)
- Vref

Sample Rate/Resolution/Frequency Response

- 48kHz, 24-bit, 20Hz – 20kHz

Ethernet

- Two Gigabit RJ-45 connections – one for AES67, one for networked remote control

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Parallel GPI/O Control Port

- 15-pin female D connector, 0-5V TTL levels; 5 inputs, 5 outputs

Front Panel Controls and Indicators

- 5-key navigation cluster; graphical color LCD display; LED status indicators for each power supply, status, and reference

Power

- Dual internal redundant auto-ranging power supplies, each rated at 100-264VAC, 50/60Hz, 100 Watts maximum

Dimensions and Weight

- 19”W x 9”D x 1.75” H (approximately 48.2 x 22.0 x 4.5 cm)
- Net weight approximately 9.0 pounds (4.08kg)
- Shipping dimensions 22”W x 20”D x 7”H (approximately 55.6 x 50.8 x 17.8 cm)
- Shipping weight approximately 16 pounds (7.26kg)

Environmental

- Operating temperature 0 to 50 degrees Celsius (32 to 122 degrees Fahrenheit)
- Non-operating temperature -20 to 70 degrees Celsius (-4 to 158 degrees Fahrenheit)

Intended Location

- Telecommunications center or dedicated computer/machine room

Regulatory

- North America – FCC and CE tested and compliant with UL-approved power supplies
- Europe – Complies with European Union Directive 2002/95/EC on the restriction of use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/EC, 2005/747/EC (RoHS directive), and WEEE

Licensing

- The Linear Acoustic LA-5300 is manufactured under license from Dolby Laboratories; Dolby, Dolby Audio, and the double-D symbol are trademarks of Dolby Laboratories.
- The NEXTGEN TV logo is an unregistered trademark of the Consumer Technology Association and is used by permission

Warranty

- Standard Telos Alliance 2-year limited parts and labor
- https://www.telosalliance.com/warranty-information